CHARGED-COUPLED DEVICES
for
ANALOG SIGNAL PROCESSING
-RECURSIVE FILTER STUDY

Veerachai Iamsa-ad

NAVAL POSTGRADUATE SCHOOL

Monterey, California

THESIS

Charge-Coupled Devices
for
Analog Signal Processing
– Recursive Filter Study

by

Veerachai Iamsa-ad

Thesis Advisor: T. F. Tao

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second-order CCD DAR filter, two MOS IC delays (SAD-100) were used for the second-order DAR filter experiments. The first-order low-pass and high-pass DAR filter experiments using the CCD and the SAD-100, and the second-order low-pass, high-pass, and band-pass DAR filter experiments using the SAD-100 only were conducted. Agreement between the experimental result and the theory is satisfactory. The result of this thesis indicates that both the CCD and the SAD-100 DAR filters may be useful for applications in which comb-filters and range gated filters are used.

Charge-Coupled Devices
for
Analog Signal Processing -- Recursive Filter Study

by

Veerachai Iamsa-ad
First Lieutenant, Royal Thai Army
B.S., The Citadel, 1968
M.S., The Naval Postgraduate School, 1974

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requirements for the degree of

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I. INTRODUCTION

Since the CCD was invented at the Bell Telephone laboratories by Drs. Boyle and Smith in 1970 [1], this new development has been receiving much attention by governmental agencies and industry because of its important properties -- low-power consumption, low noise, and delay capability. Many potential applications have been investigated such as imaging, memory, and signal processing, and reports of success were frequently published.

In the signal processing field, both digital and analog approaches are possible. Although digital signal processing is a fast growing concept which offers improved signal-to-noise ratio, accuracy and flexibility, there are areas in which analog signal processing will not be replaced in the near future. In a system, these are the points at which the signal-to-noise ratio varies over a wide dynamic range; therefore, if the signal is to be converted from analog to digital, it must have many bits of information per sample which is too high for the present analog-to-digital converters to cope with. One example is the front-end of a receiving system where the signal may vary over a wide dynamic range. Usually analog-to-digital conversion is done at a point in the system where the dynamic range is smaller.

Many electronics companies have been experimenting in CCD signal processing, especially in transversal or nonrecursive filtering. The objective of this thesis was directed toward recursive filtering using CCD. There are advantages that recursive filtering has over nonrecursive filtering. In the frequency domain design, a recursive filter, having infinite memory, can reproduce a desired frequency response with fewer delays than a nonrecursive filter, which has finite memory. Besides, recursive filtering offers a feasibility of using building block approach which will facilitate filter designing.

A. DIGITAL APPLICATIONS

Since the CCD's are true analog sampled-data devices, there is a possibility of an analog and digital version of the same function in application of this technique to signal processing systems. Digital designs offer several advantages such as the inherent good noise margin and flexibility in providing many functions with one design. Besides, the present CCD technology is sufficient to ensure the proper operations of the functions designed [2].

The TRW System Group [2] proposed that it was possible and beneficial to design a general purpose CCD arithmatic chip which would serve as the building block for a number of digital designs. This CCD arithmatic chip would contain four 16-bit multipliers, three 16-bit adders, two multiplexers, and gating and delay operations, all of which fabricated on a 300 mils by 350 mils chip. The object of this configuration was to perform the kernel operation of the fast fourier transformation. There would be other functions that could be provided by this chip. Some of the possibilities are an inverse fast fourier transformer, a recursive digital resonator, a lattice digital filter, a serial correlator, or an output summer for a CCD correlator. Other configurations were also proposed such as a 32-stage correlator chip and a fast fourier transform memory chip.

With the mentioned configurations, the TRW System Group indicated that possible applications included the following systems: radar range-gated and combination filters, sonar range-gated and combination filters, guidance and and navigation Kalman filter, communications, and voice processing.

B. ANALOG APPLICATIONS

Many laboratories have been working with transversal filters, matched filters, and correlators. In these applications, the CCD is nondestructively tapped at equidistant positions along the array. These output signals are then weighted according to the design and summed to give the desired characteristics. Applications for these filters are in radar, sonar, voice processing, spread spectrum communication, and many others.

In signal processing, CCD filters may be very useful in the area of frequency filtering. As mentioned, a recursive filter has advantages over a nonrecursive one in frequency-domain design. The explanation of how a CCD can be used to implement a recursive filter is presented below.

If a low-pass filter comparable in performance to a passive RC filter as shown in Fig. 1.1 is to be implemented, the approach can be analyzed as follows [2].

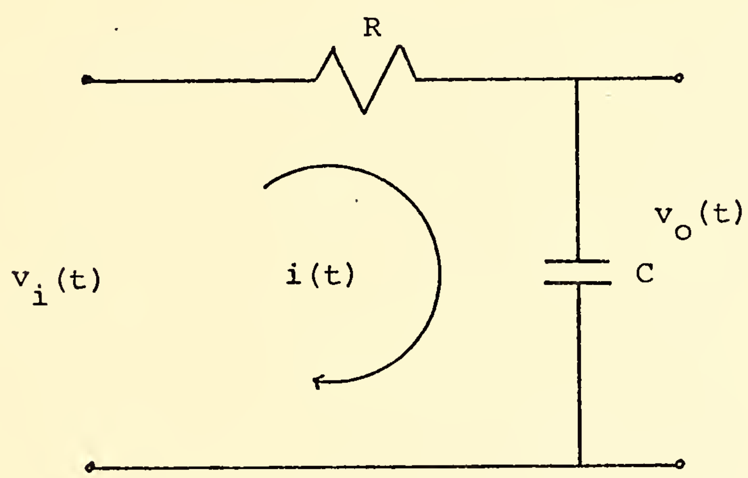

Figure 1.1 RC Low-pass Filter

$$C \frac{d}{dt} v_o(t) = \frac{v_i(t) - v_o(t)}{R} \qquad (1.1)$$

Equation 3.1 can be represented by a difference equation as follows.

$$v_o(n) = \frac{1}{1+K} v_i(n) + \frac{K}{1+K} v_o(n-1) \qquad (1.2)$$

where $K = \frac{RC}{T}$

$$= \frac{1}{2\pi f_L T}$$

$v_i(t)$ and $v_o(t)$ are the time-varying voltage input and output, respectively, and $v_i(n)$ and $v_o(n)$ are the discrete voltage input and output, respectively, at time n. The cut-off frequency of the filter is f_L. The output of the filter, as described by equation 1.2, can be obtained by recirculating and adding the output of the previous time frame to the input signal of the present time frame, with proper weighting coefficients. CCD, with its inherent delay capibility, can provide the delay and, in addition, other hardware needed is for analog addition and multiplication. Further simplification can be obtained by using ratio of resistances for weighting coefficients. Thus, a CCD discrete analog filter is possible.

An analysis of a high-pass filter given below is parallel to that of the low-pass filter example previously explained. Figure 1.2 shows a passive RC high-pass filter.

$$\frac{v_o(t)}{R} = C \frac{d}{dt}\{v_i(t) - v_o(t)\}$$

$$\frac{v_o(n)}{R} = \frac{C}{T}[\{v_i(n) - v_o(n)\} - \{v_i(n-1) - v_o(n-1)\}]$$

$$v_o(n) = \frac{1}{1+K} [\ v_i(n) - v_i(n-1) + v_o(n-1) \ ] \qquad (1.3)$$

where $K = \dfrac{RC}{T}$

$$= \frac{1}{2\pi f_H T}$$

and f_H is the cut-off frequency of the filter.

Again, CCD can be used to implement a high-pass filter.

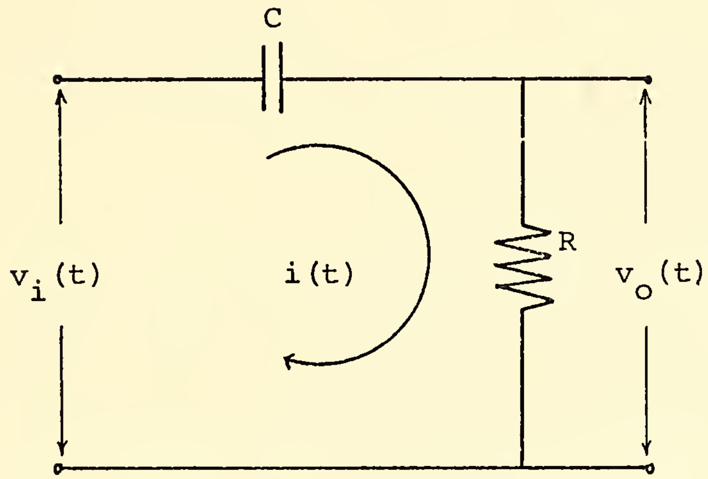

Figure 1.2 RC High-pass Filter

II. THEORIES OF THE CCD

A charged-coupled device is basically an array of metal-insulator-semiconductor (MIS) capacitors. The materials used are usually Al-SiO_2-Si. Other semiconductors can be chosen to suit a special application. For example, InSb MIS could be used for infrared detection applications. The basic operation of a CCD can be thought of as that of an analog shift register in which the signal is represented by minority carriers in the depleted regions, or potential wells, created in the semiconductor at the insulator-semiconductor interface of the MIS capacitor. The voltages on the metal gates can be manipulated to transfer the signal, unidirectionally, from one potential well to the next.

The structural diagram of a 9 bit CCD* used in this thesis is shown in Fig. 2.1. Note that a signal processed by this CCD will be delayed by 9 unit-delays. One unit-delay equals the inverse of the clock frequency.

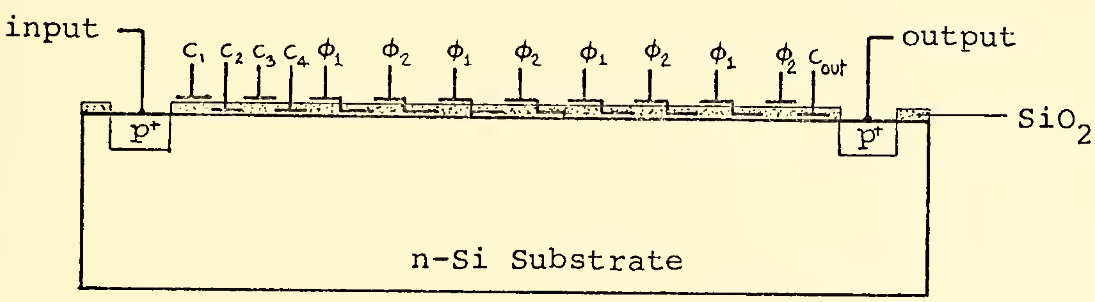

Figure 2.1 Structural Diagram of the 9 bit CCD

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* Courtesy of Microelectronics Center, TRW Systems Group, Redondo Beach, Calif.

A• PRINCIPLES OF OPERATION

1. Storage

Figure 2.2 shows a p-channel MIS capacitor which is the basic element of a CCD. and Fig. 2.3 shows the C-V and the C-t characteristics of such a device [3].

When voltage on the metal gate is positive, the MIS is in the accumulation region. As the voltage is quickly changed from postive to negative, the semiconductor becomes depleted exposing the positive, immobile ions until the dark current brings it to the high frequency inversion region. This process takes place within the storage time of the semiconductor. A CCD needs to remain in deep depletion long enough to process the signal and, therefore, the semiconductor which is used to fabricate CCD usually has long time constant.

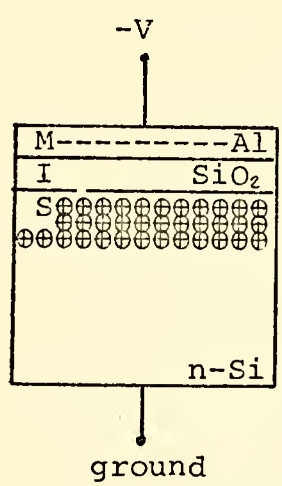


Figure 2.2 MIS Capacitor

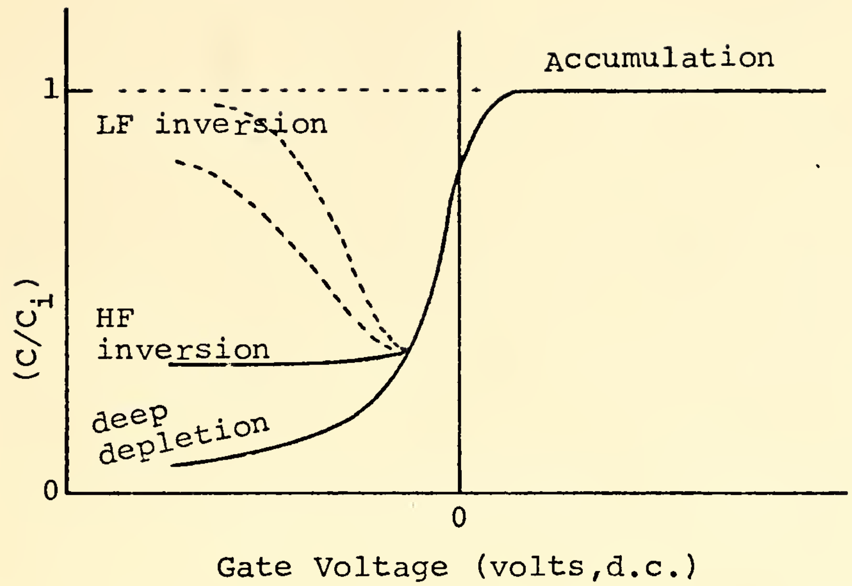


Figure 2.3 (a) C-V Characteristics of a MIS Capacitor

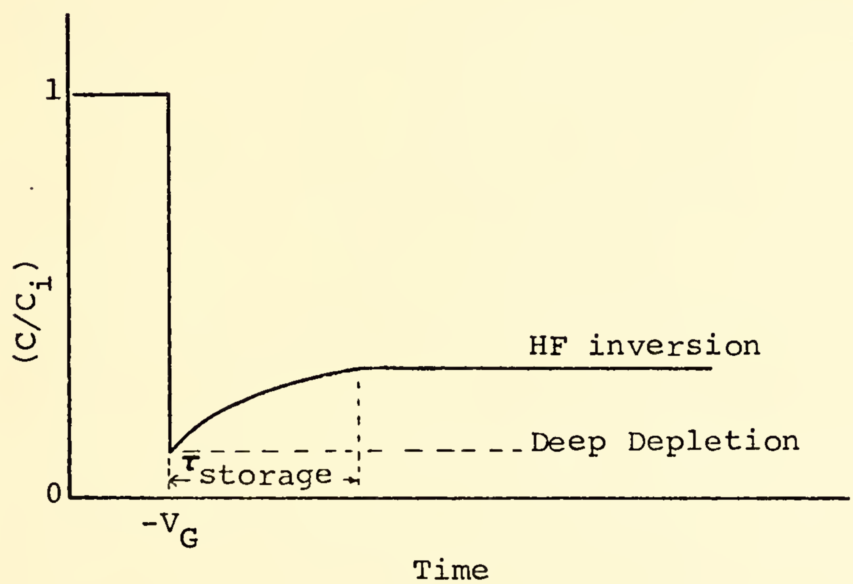


Figure 2.3 (b) C-t Characteristics of a MIS Capacitor

2. Transfer Process

A CCD which consists of an array of MIS capacitors must be operated with two or more sets of clocks applied to its gates. Classification of CCD's is based on the number of clock waveforms used. The more common CCD's are two-phase, three-phase, and four-phase [1].

The clock waveforms used with the CCD in this thesis project are shown in Fig. 2.4, along with illustrations of how a signal, represented by minority carriers, can be unidirectionally shifted from the input to the output. At time t_1, the potential wells under the gates connected to $\phi 2$ are the deepest, thus charges which represent the signal are collected under these gates. At time t_2, these charges are transferred to the next gate on the right because of the transition in levels of $\phi 1$ and $\phi 2$.

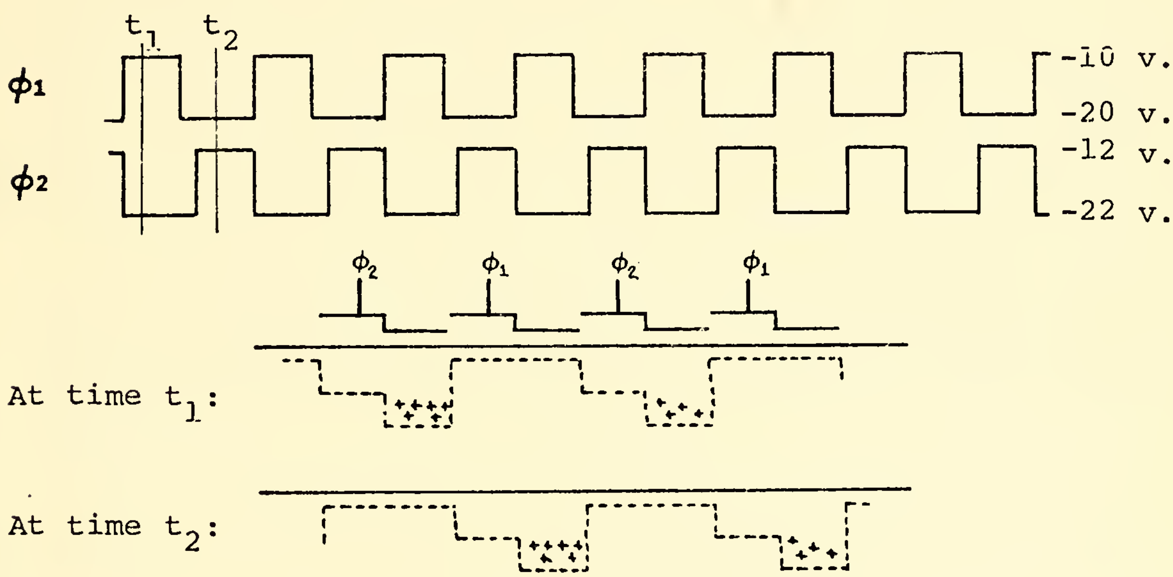


Figure 2.4 Clock Waveforms for the 9 bit CCD.

3. Input Techniques

For CCD's as described, there are a number of input techniques used by various experimenters such as electrical serial input, optical input, and electrical parallel input [1]. The sloshing technique used in this thesis project is illustrated in Fig. 2.5, which also shows the relationship of the input clock and the ø2 clock.

For the operation, voltages on the c_1 and c_2 gates are held constant to produce the surface potential distribution as illustrated. At time t_1, the input diode is forward biased by the input clock. At the same time, the ø2 clock is high and produce no depletion region under the first clock gate; therefore, minority carriers will be supplied to the potential wells under the c_1 and c_2 gates. At time t_2, the input diode is reverse-biased and the excess minority carriers are drawn out leaving only a number corresponding to the difference of the voltages on the c_1 and c_2 gates. at time t_3 and t_4, shifting operation begins. It is clear that any level of signal or zero reference (fat zero) can be represented by the difference between the quiescent voltages on gate c_1 and gate c_2. The signals can be modulated onto either one or both of the gates. A careful consideration will show that with respect to the input, modulation on gate c_1 will yield in-phase output while modulation on gate c_2 will yield 180° phase-shifted output.

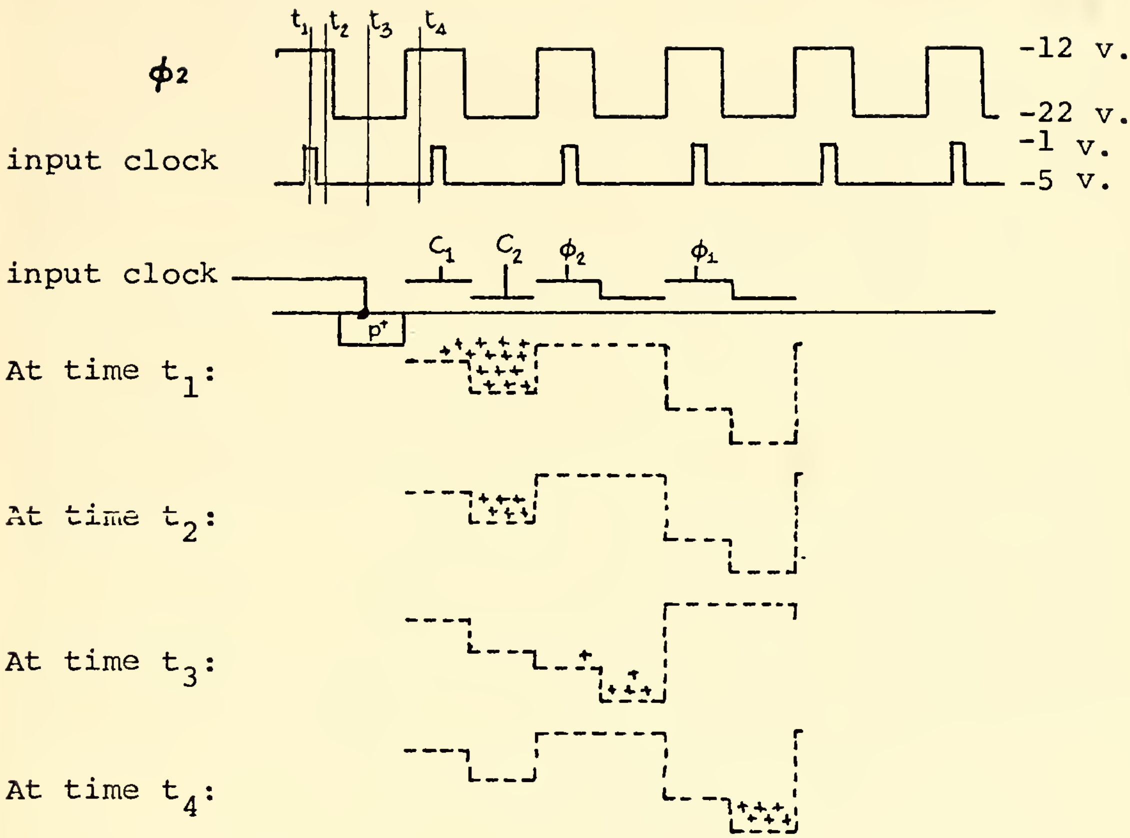


Figure 2.5 Sloshing Input Configuration.

4. Output Techniques

After the input signal is transferred along a CCD, the output is obtained at the output diode. Reference 1. discussed three methods of detecting the signal at the output, injection of charges into substrate, reverse biasing of pn junction, and floating gate scheme. A diagram in Fig.2.6 shows the technique used with the CCD's used in this thesis project.

The voltage on the output gate is always held constant at approximately -30v. The signal charges will be dumped onto the output diode when the ø2 clock is highly negative. The function of the refresh amplifier is to be the sink for the signal charges from the previous packet and reverse bias the output diode immediately before the next packet arrives; this is accomplished by the refresh pulse at the moment the ø2 clock goes high. After refresh, when the signal charges come, the output current of the output source follower will follow the signal, faithfully.

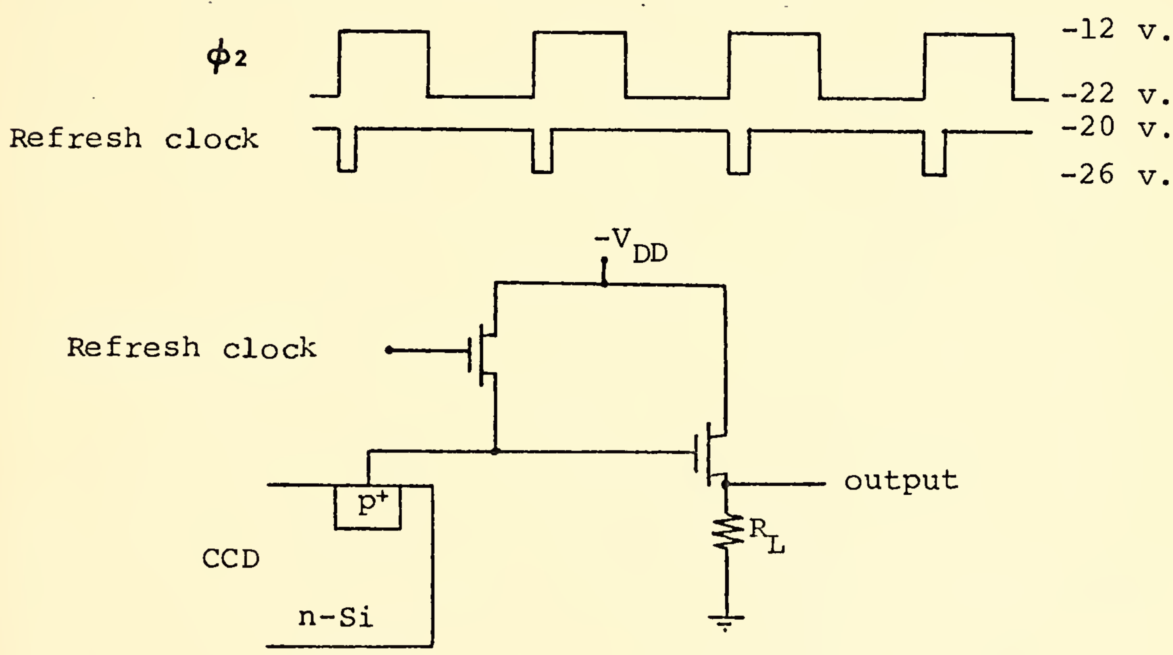


Figure 2.6 Output Circuit Configuration

B. NOISE IN CCD

References 4 and 5 give a very thorough treatment of this subject. The CCD is a very low-noise device. For a surface channel CCD having 1000 gates, the noise caused by the device itself is typically less than 1000 electrons. The sources of noise in a CCD are the background-charge generation, transfer fluctuation, fast-interface state loss, and output amplifier.

III. THEORETICAL CONSIDERATIONS OF THE CCD RECURSIVE FILTER

A. USE OF ANALOG PROTOTYPES FOR CCD RECURSIVE FILTERS

The transfer functions of classical continuous signal filters, both passive filters and active filters, are expressed in the s-domain. If specific characteristics of a desired filter are known, a transfer function can be selected from one of many families of well known classical filters. One of these families is the Chebychev family which consists of the Butterworth filter, the elliptic filter, and the Chebychev filter. The Guassian family includes the Bessel filter, the Paynter filter, and the Guassian filter. This thesis will include investigations of using the analog prototypes from the Butterworth filters family for CCD recursive filters.

1. Butterworth Filters

The design procedures of a CCD filter starts with a low-pass one. Typical frequency responses of Butterworth filters are shown in Fig. 3.1. The distinct property of a Butterworth filter is that its frequency response is monotonic in both the passband and the stopband and is maximally flat at zero frequency.

The transfer function, H(s), of a Butterworth low-pass filter has a constant numerator. Its denominator is determined by the poles on the circumference of a circle of radius $2\pi f_L$ in the s-plane, where f_L is the cut-off frequency of the filter. Only the poles in the left half of the s-plane are considered. Figure 3.2 gives details of the pole locations and transfer functions of the first-order and the second-order Butterworth filters[6].

Only the first-order and the second-order filters will be discussed for the reason which is given in III.B.3.

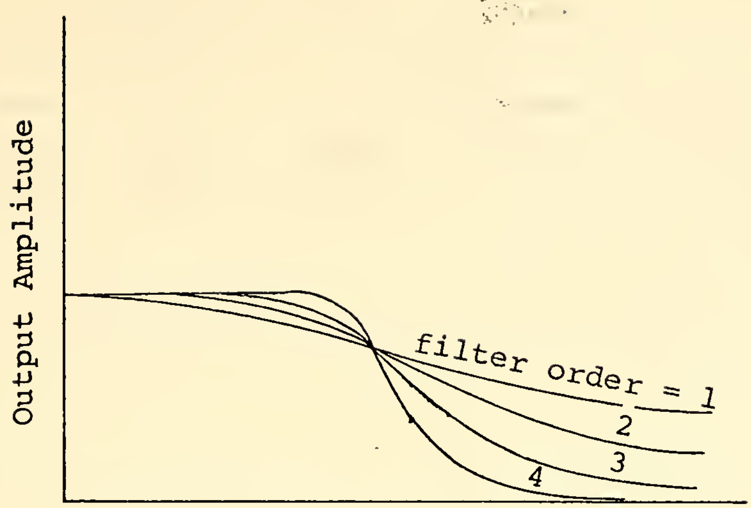
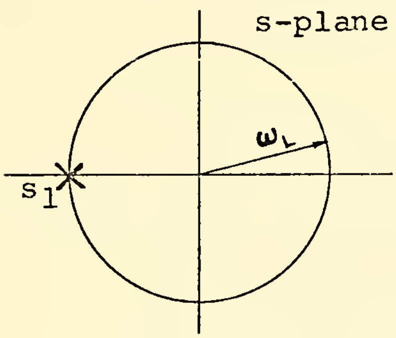
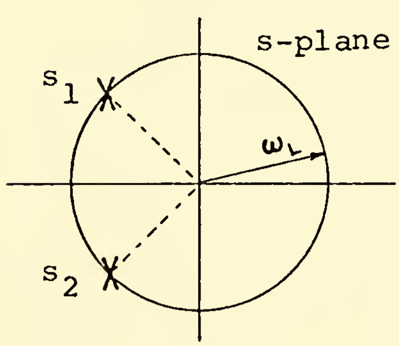


Figure 3.1 Frequency Response of Low-pass Butterworth Filters



The first-order transfer function is

$$H(s) = \frac{s_1}{s - s_1} = \frac{\omega_L}{s + \omega_L}.$$



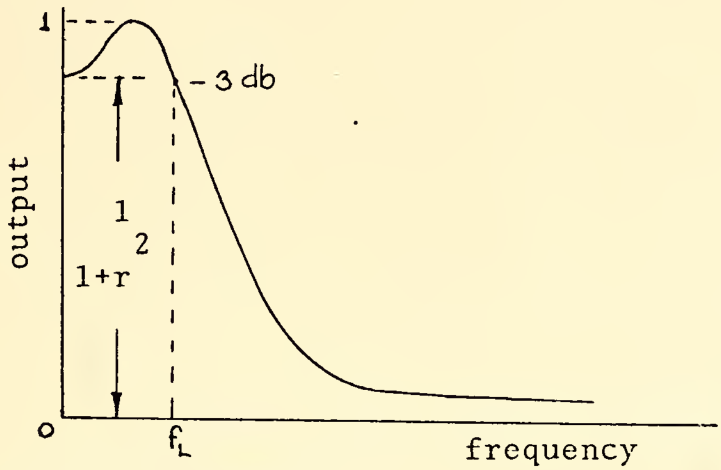
The second-order transfer function is

$$H(s) = \frac{s_1 s_2}{(s-s_1)(s-s_2)} = \frac{\omega_L^2}{\left\{s+\frac{\omega_L}{2}\right\}^2 + \left\{\frac{\omega_L}{2}\right\}^2}.$$

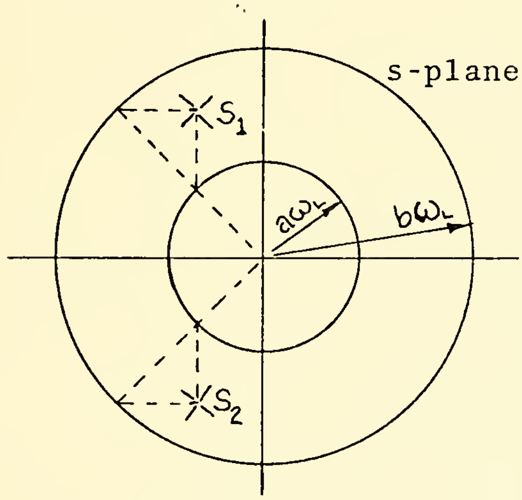
Figure 3.2 Pole locations of Low-pass Butterworth Filters

2. Chebychev Filters

A Chebychev filter has the fastest roll-off of any all-pole designs. It is monotonic in the stopband and has constant ripple in the passband. More detailed design procedure can be found in Ref. 6. Figure3.3 shows the frequency response and pole locations of Chebychev filters.



(a) Frequency Response



The second-order transfer function is

$$H(s) = \frac{s_1 s_2}{(s-s_1)(s-s_2)}$$

$$b,a = \frac{1}{2}\left[\left\{\sqrt{r^{-2}+1} + r^{-1}\right\}^{1/2} \pm \left\{\sqrt{r^{-2}+1} + r^{-1}\right\}^{-1/2}\right]$$

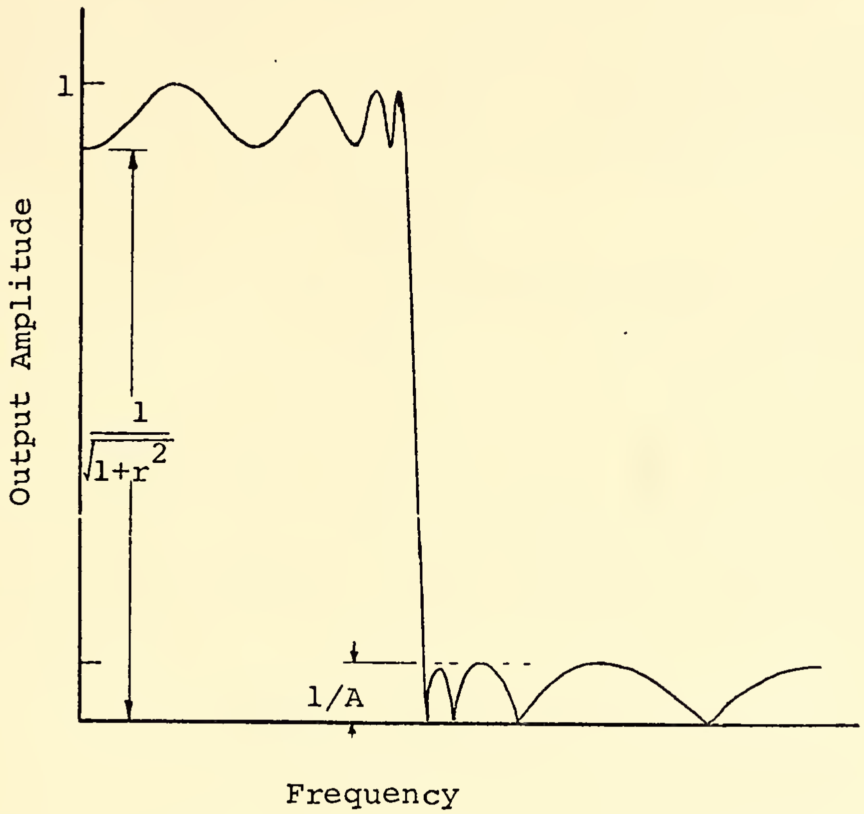
where r is the ripple factor.

(b) Pole locations.

Figure 3.3 Frequency Response and Pole Locations of Second-order Chebychev Filters

3. <u>Elliptic Filters</u>

An elliptic filter offers the fastest roll-off of any kind of filters but has ripples in both the passband and the stopband. Figure 4.4 shows frequency response of such a filter. Reference 6 has detailed discussion on this



r is the ripple factor and A is the attenuation factor. The order of the filter is 8.

Figure 3.4 Frequency Response of an Elliptic Filter

B. DESIGN OF CCD DISCRETE ANALOG RECURSIVE FILTER

1. Transformations

Since a charged-coupled device is an analog shift register in nature and its input is sampled, a filter implemented by CCD is a discrete analog filter. As shown in I.B., a CCD filter utilizes delay as its mechanism of filtering as in digital filter ; therefore, in designing a CCD recursive filter , a transformation of the specified transfer function in s-domain to z-domain is needed. Two methods of transformations will be discussed.

a. Standard Z-Transform

Firstly, the definition of z is

$$z = e^{sT} \tag{3.1}$$

where T is the delay period.

When the specifications of a filter are determined, its transfer function in s-domain is obtained, then the transformation into z-domain can be done by two methods which are given below.

(1) Impulse Response Method

When H(s) is obtained, the impulse response of the filter is then determined by inverse Laplace transformation. If

$$h(t) = \text{impulse response of the filter.}$$

and

$$h^*(t) = \text{sampled impulse response}$$

$$= \sum_{n=0}^{\infty} h(nT) \cdot \delta(t-nT) \quad , \; n = 0,1,2,\ldots$$

From Laplace transformation,

$$\mathcal{L}[h^*(t)] = \sum_{n=0}^{\infty} h(nT) \cdot e^{-nsT}$$

then, $$H(z) = \sum_{n=0}^{\infty} h(nT) \cdot z^{-n} \tag{3.2}$$

which is the desired transfer function in the z-domain.

There is an important property of the standard z-transform which must be noted at this point. The Laplace transform of h(t) implies that if

$H_i(s)$ transforms into $H_i(z)$,

then $H_1(s) + H_2(s) + \ldots$ transform into $H_1(z) + H_2(z) + \ldots$

and $H_1(s) \cdot H_2(s) \ldots$ do not transform into $H_1(z) \cdot H_2(z) \ldots$

(2) Z-Transform Tables

There exist z-transform tables in most of the discrete system textbooks. These tables can be used to determined H(z) when H(s) and/or h(t) is known.

b. Bilinear Z-Transform

Reference 7 indicates that aliasing, the recurrence of transfer characteristics with respect to frequency, is an important problem with the standard z-transform method. Consequently, the resulted discrete analog filter can be used with only the signals whose Nyquist frequencies are much smaller than one-half of the effective sampling frequency of the CCD (the effective sampling frequency is the CCD clock frequency divided by the number of bits of the CCD).

The bilinear z-transform is treated very thoroughly in Ref. 6 . The transform is described by the relationship

$$s = \frac{2}{T} \cdot \frac{z-1}{z+1} \tag{3.3}$$

which indicates mapping the entire left-half of the s-plane into the interior of the unit circle in the z-plane. This always gives one or more zeroes at z = -1, or at one-half of

the effective sampling frequency ; therefore, aliasing problem is eliminated.

This transformation is particularly suited for a filter that is required to have flat magnitude-response in the passband and the stopband. However, there exists the nonlinear warping of the frequency scale in the transformation. The frequencies in the z-palne is related to the frequencies in the s-plane by

$$\frac{\omega_p T}{2} = \tan \frac{\omega_z T}{2}, \qquad \text{or}$$

$$f_p = (1/\pi T)\bullet\tan(\pi f_z T), \tag{3.4}$$

where $f_p = \omega_p'/2\pi$ = cut-off frequency of the analog prototype, and $f_z = \omega_z/2\pi$ = desired cut-off frequency of the discrete analog filter.

Then, if the desired cut-off frequency of the filter is f_z, the compensated frequency, f_p, must be used in determining H(s) of the analog prototype.

2. Implementation

When H(z) of the desired discrete analog filter has been obtained, different methods of implementation mostly follow those of digital filters which are discussed in Ref. 6. They are presented here for completeness.

a. Direct Form

Suppose that

$$H(z) = \frac{a_0 + a_1 z^{-1} + a_2 z^{-2} + \ldots + a_N z^{-N}}{1 + b_1 z^{-1} + b_2 z^{-2} + \ldots + b_M z^{-M}}, \qquad N \leq M. \tag{3.5}$$

Figure 3.5 shows the direct form implementation of the

filter. Each z^{-1} block represents one unit-delay. The direct form is not the way one would choose to implement a discrete analog filter because too many delays are used. However, it provides insight as to how a filter can be implemented from a transfer function.

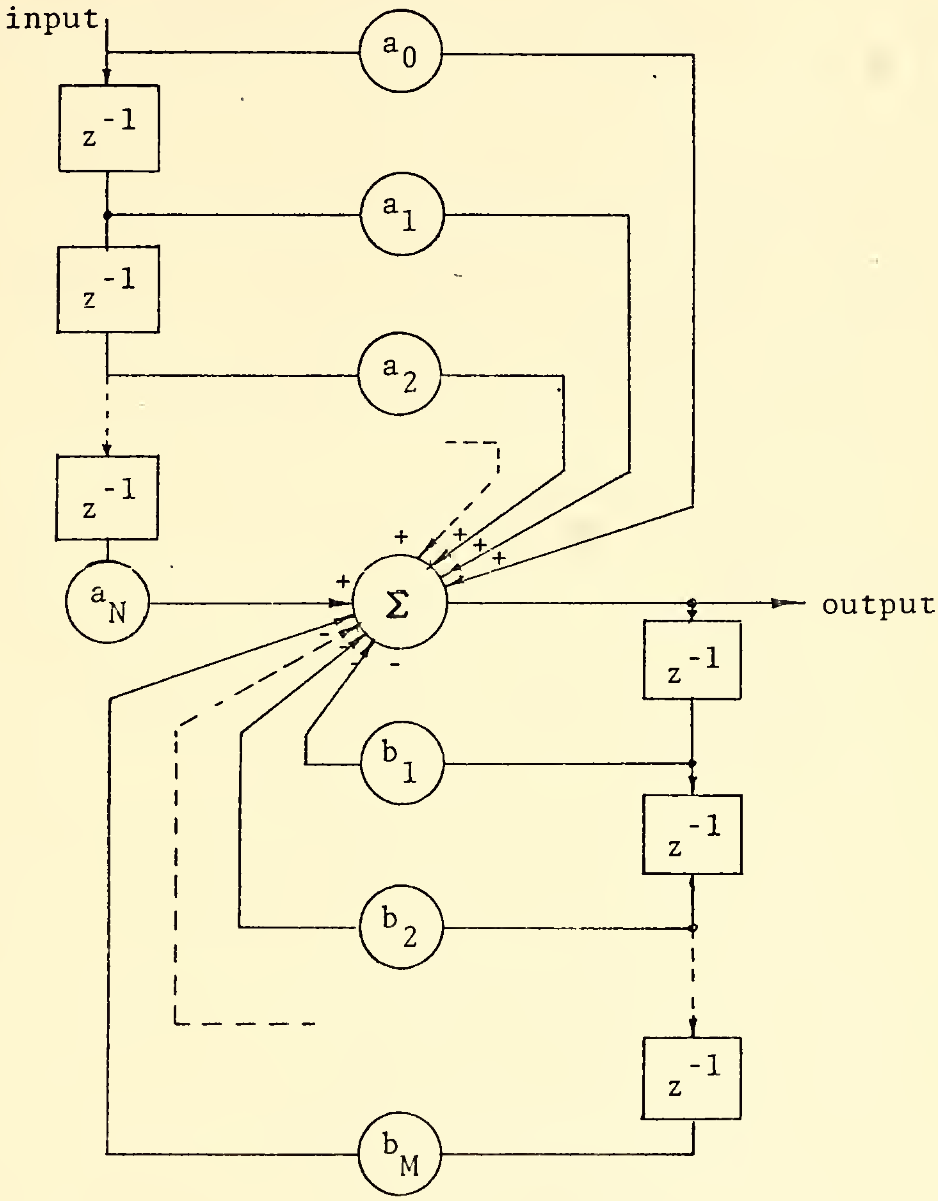


Figure 3.5 Direct Form implementation

b. Canonic Form

The second form of implementation is called the canonic form. The The transfer function for this form is the same as that of the direct form[6].

$$H(z) = \frac{a_0 + a_1 z^{-1} + a_2 z^{-2} + \ldots + a_N z^{-N}}{1 + b_1 z^{-1} + b_2 z^{-2} + \ldots + b_M z^{-M}}, \qquad N \leq M. \tag{3.6}$$

Figure 3.6 shows the canonic form implementation of the filter. This form uses as many delay blocks as there are coefficients in the denominator of the transfer function; therefore, a reduction in the number of delays is achieved which can be as many as one-half of that needed for the direct form is obtained.

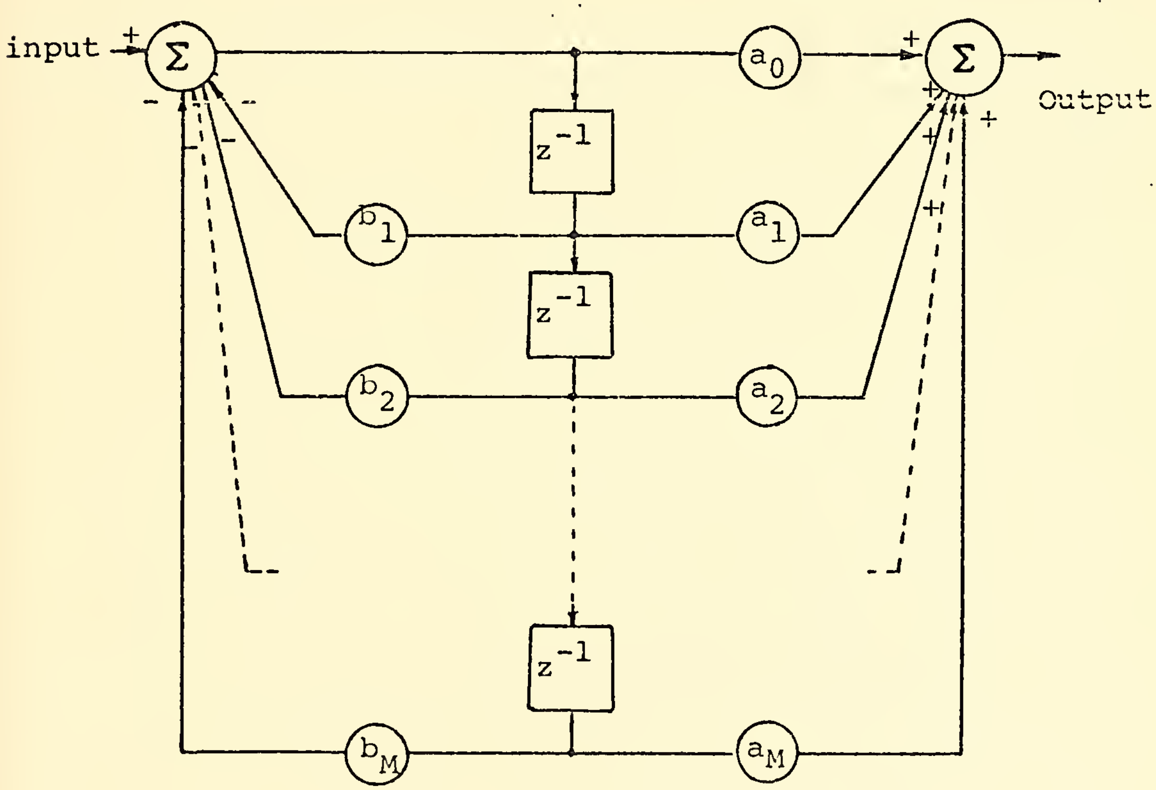


Figure 3.6 Canonic Form Implementation

c. Cascade Form

The third form of implementation is the cascade form. The transfer function, H(z), is factored into a product of low-order funtions -- preferably, first-order or second-order.

$$H(z) = \prod_{m=1}^{M} \left\{ \frac{a_{m0} + a_{m1} z^{-1} + a_{m2} z^{-2}}{1 + b_{m1} z^{-1} + b_{m2} z^{-2}} \right\} . \qquad (3.7)$$

Each factor of H(z) can then be individually implemented by the canonic form. These low-order blocks can then be cascaded to give the desired filter. Figure 3.7 shows the cascade form implementation[6].

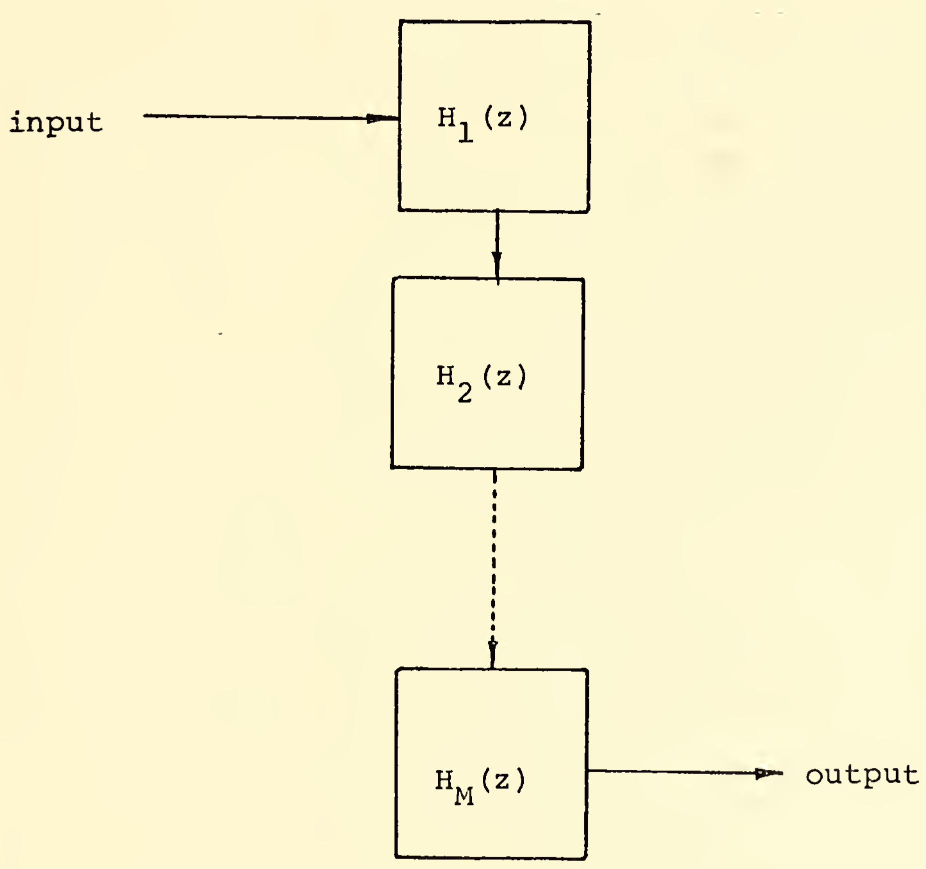


Figure 3.7 Cascade Form Implementation.

d. Parallel Form

The fourth form of implementation is the parallel form. If the transfer function is in partial-fraction form,

$$H(z) = \sum_{m=1}^{M} \left\{ \frac{a_{m0} + a_{m1} z^{-1} + a_{m2} z^{-2}}{1 + b_{m1} z^{-1} + b_{m2} z^{-2}} \right\}. \tag{3.8}$$

low-order blocks can be implemented in the canonic form and then connected in parallel as shown in Fig. 3.8.

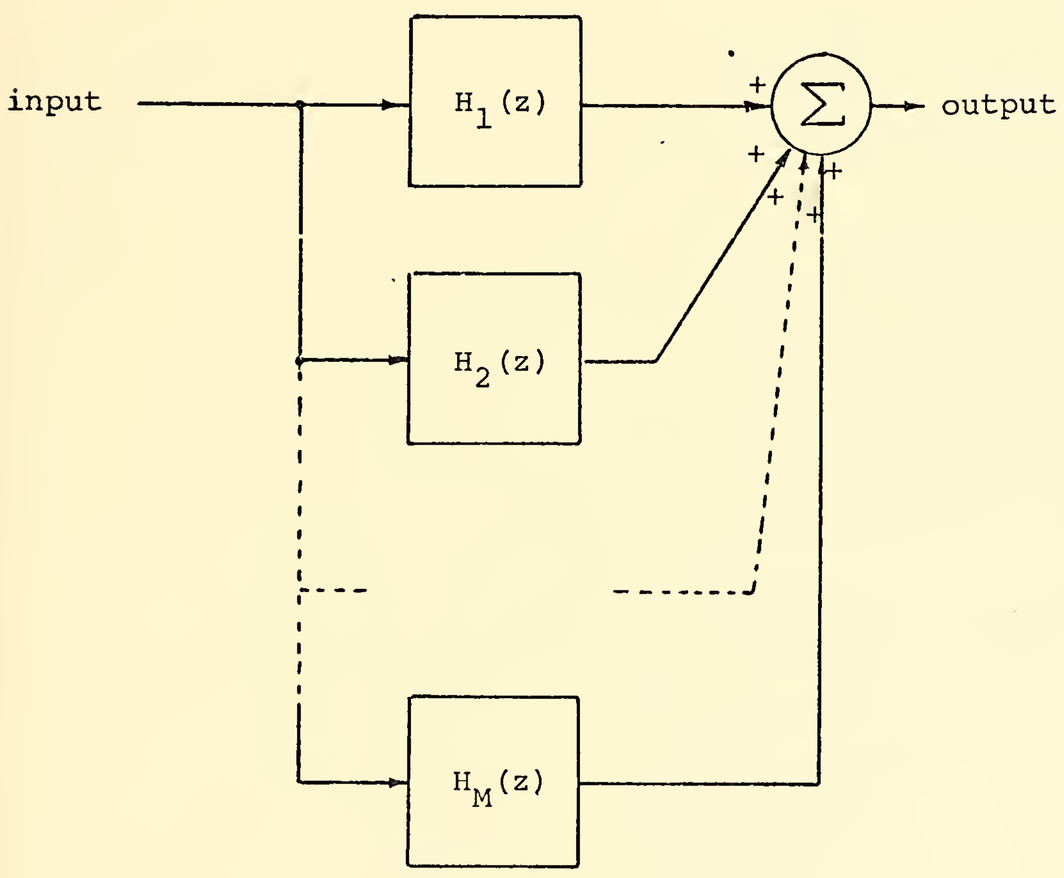


Figure 3.8 Parallel Form Implementation.

3. Building Block Approach

The CCD discrete analog recursive filters can be very useful in real time signal processing. To optimize their flexibility and versatility, the design must be in some basic form which is easily applied to many applications. A second-order filter building block design will provide this versatility. Its transfer function is in the following form:

$$H(z) = \frac{a_0 + a_1 z^{-1} + a_2 z^{-2}}{1 + b_1 z^{-1} + b_2 z^{-2}} \, . \qquad (3.9)$$

The implementation of this second-order discrete analog filter is in canonic form as shown in Fig. 3.9. One can see immediately that a first-order filter can be obtained from this building block by simply setting the second-order coefficients to zero. Two or more sections can be cascaded or connected in parallel to give more complex filters.

Next, an example is given to illustrate a second-order Butterworth type low-pass CCD discrete analog recursive filter. Both the standard z-transform and the bilinear z-transform methods are used, and relationships between the filter cut-off frequencies and coefficients are derived below.

From Fig. 3.2, the transfer function of a continuous low-pass Butterworth filter is

$$H(s) = \frac{\omega_L^2}{\left(s + \frac{\omega_L}{2}\right)^2 + \left(\frac{\omega_L}{2}\right)^2} \, , \qquad (3.10)$$

where ω_L = critical radian frequency.

If $k = \frac{\omega_L}{\sqrt{2}}$,

then $$H(s) = \frac{2k^2}{(s+k)^2 + k^2}.$$

By the standard z-transform [8]:

$$H(z) = \frac{2ke^{-kT}\sin kTz^{-1}}{1 - 2e^{-kT}\cos kTz^{-1} + e^{-2kT}z^{-2}}.$$

By the bilinear z-transform:

Let $k' = \frac{\omega_L T}{2\sqrt{2}}$,

then $$H(z) = \frac{\omega_L^2}{\frac{2}{T}\cdot\frac{z-1}{z+1} + \frac{\omega_L}{2}^{2} + \frac{\omega_L}{2}^{2}}$$

$$= \frac{2k'^2}{\frac{z-1}{z+1} + k'^{\,2} + k'^{\,2}}$$

$$= \frac{\frac{2k'^2}{2k'^2+2k'+1}(1+2z^{-1}+z^{-2})}{1 + \frac{4k'^2-2}{2k'^2+2k'+1}z^{-1} + \frac{2k'^2-2k'+1}{2k'^2+2k'+1}z^{-2}}.$$

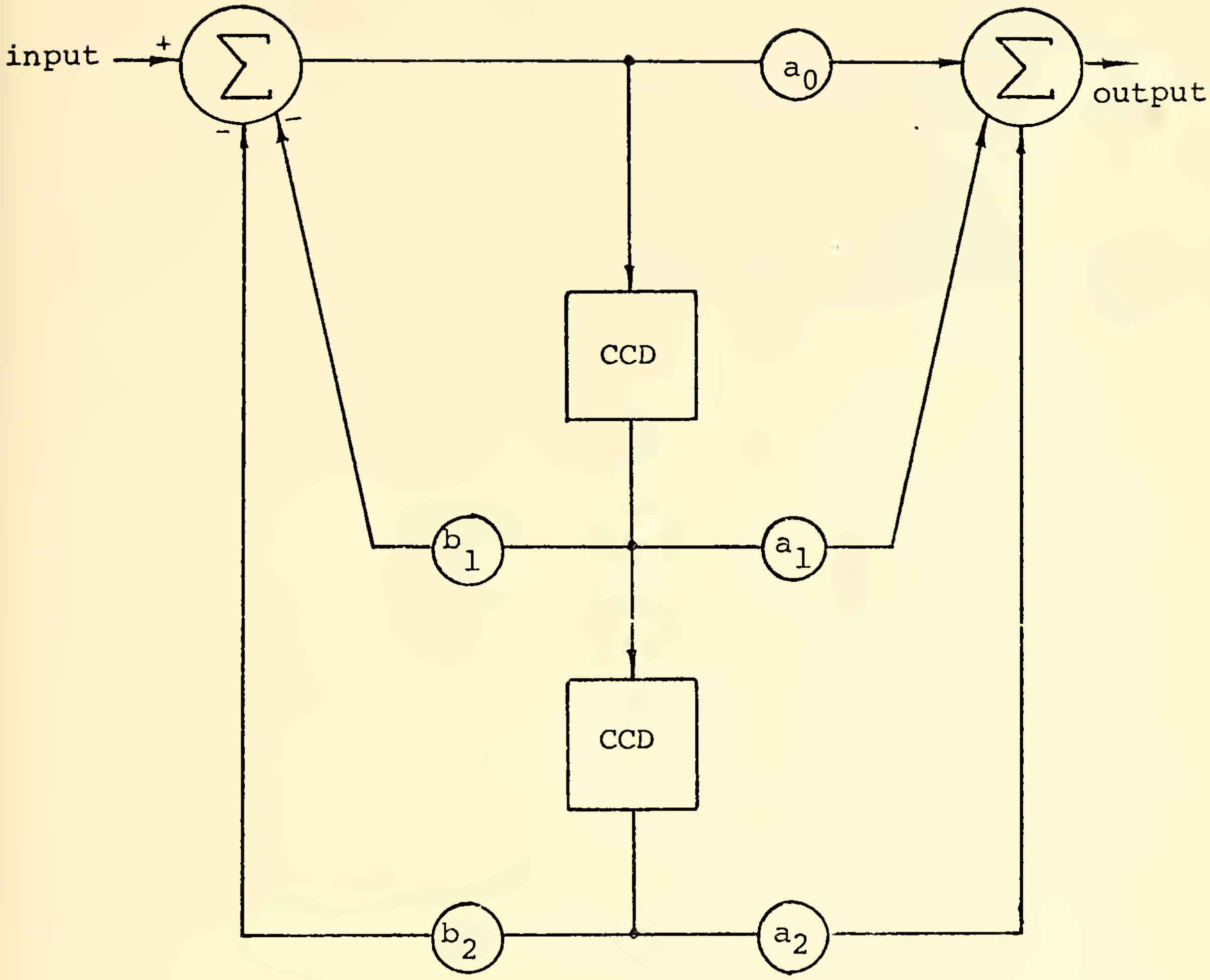


Figure 3.9 Block Diagram of a Second-order CCD Discrete Analog Filter

For the first-order low-pass CCD discrete analog recursive filter, whose prototype transfer function is shown in Figure 3.2, the table of coefficients is shown in table I. The coefficients of a second-order Butterworth low-pass CCD discrete analog recursive filter can be constructed as shown in table II.

Table I Coefficients for First-order Butterworth Low-pass Discrete Analog Recursive Filter

	a_0	a_1	a_2	b_1	b_2
Standard z-transform	ω_L	0	0	$-e^{-\omega_L T}$	0
Bilinear z-transform	$\frac{\omega_p T}{2+\omega_p T}$	$\frac{\omega_p T}{2+\omega_p T}$	0	$-\frac{2-\omega_p T}{2+\omega_p T}$	0

Table II Coefficients for Second-order Butterworth Low-pass Discrete Analog Recursive Filter

	a_0	a_1	a_2	b_1	b_2
Standard z-transform	0	$2ke^{-kT}\sin kT$	0	$-2e^{-kT}\cos kT$	e^{-2kT}
Bilinear z-transform	$\frac{2K'^2}{2K'^2+2K'+1}$	$\frac{4K'^2}{2K'^2+2K'+1}$	$\frac{2K'^2}{2K'^2+2K'+1}$	$\frac{4K'^2-2}{2K'^2+2K'+1}$	$\frac{2K'^2-2K'+1}{2K'^2+2K'+1}$

C. SAMPLING EFFECT ON CCD DISCRETE ANALOG RECURSIVE FILTER

Reference 7 indicates that a signal, f(t), when sampled can be represented by

$$f(t) = \sum_{n=0}^{\infty} f(nT) \cdot z^{-n}. \tag{3.11}$$

From the definition of z which is described by equation 3.1, it can be shown that

$$z = e^{j2\pi(f/f_c)}, \tag{3.12}$$

where f is the instantaneous frequency and f_c is the clock frequency. Therefore, the frequency spectrum of the sampled signal will be periodic in frequency with period f_c. The signal is adequately represented by the frequency spectrum of the sampled signal within the bandwidth of 0 Hz to $f_c/2$ Hz, and can be recovered by an ideal low-pass filter of the same bandwidth.

It is clear from the theory of charged-coupled device that a signal processed by a CCD is a sampled signal with a sampling period, T, which is the inverse of the CCD clock frequency, f_c. Therefore, the sampling concept explained above is applicable and any signal to be processed by charged-coupled devices must then be bandlimited to within $f_c/2$ Hz to avoid aliasing effect.

If the CCD used in a discrete analog recursive filter is one-bit long, there is no complication as far as the spectrum of the processed signal and the frequency response of the filter are concerned. The processed signal, if bandlimited to $f_c/2$ Hz can be recovered by an ideal low-pass filter of the same bandwidth. However, if a CCD is n-bit long and the filter is of the recursive type, the effective sampling frequency of the filter is not the same as the CCD clock frequency. If f_s is the effective sampling

frequency, then

$$f_s = \frac{f_c}{n} \qquad (3.13)$$

because the signal is fed back every n/f_c second.

The effective sampling period then is the inverse of f_s which is the period to be used in the unit-delay operator, z^{-1}, in the transfer function, $H(z)$.

The frequency response of the discrete analog recursive filter can then be analyzed from its unit-impulse response.

$$H(z) = \frac{V_o(z)}{V_i(z)},$$

where $V_i(z)$ is the z-transform of the unit-impulse input. Since the z-transform of a unit-impulse signal is

$$V_i(z) = 1.$$

The output spectrum, or the frequency response of the filter, is a function of z^{-1} and

$$\left|H(j\omega)\right| = \left|f(e^{-j\omega T})\right|. \qquad (3.14)$$

For example, consider a low-pass filter with the following transfer function,

$$H(z) = \frac{1}{1 + b_1 z^{-1}},$$

where $b_1 = -e^{-\omega_L T}$

which is to be implemented by a 9 bit CCD with clock frequency = 67.50 kHz

Therefore, $f_s = 7.5$ kHz

If the desired critical frequency is 500 Hz,

$$b_1 = -\ 0.658$$

The magnitude-frequency response of the filter is

$$\left|H(jf)\right| = \left|\frac{1}{1 - 0.658e^{-j2\pi(f/7500)}}\right|,$$

which is plotted in Fig. 3.10.

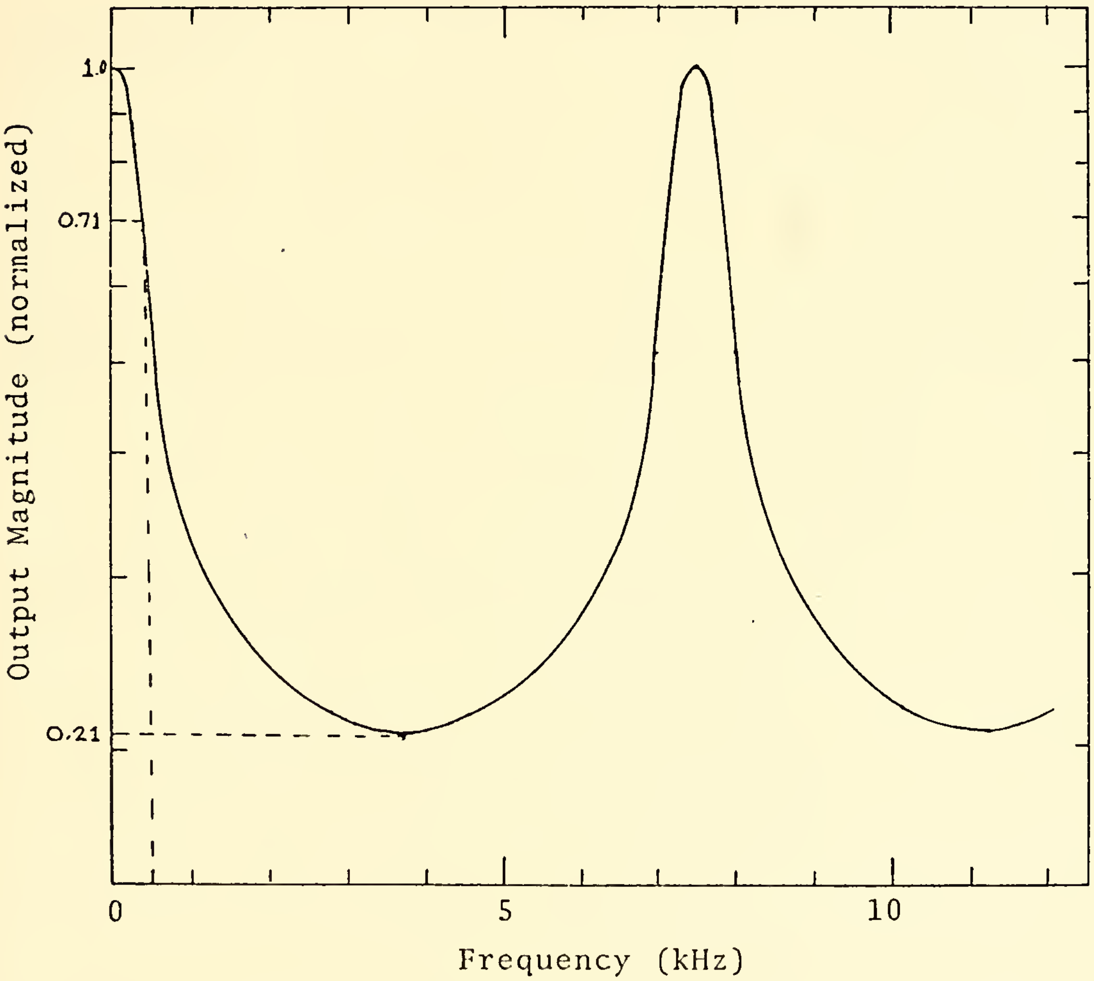


Figure 3.10 Frequency Response of a 9 bit CCD Low-pass Filter

D. TIME RESPONSE OF DISCRETE ANALOG RECURSIVE FILTER

The voltage time response to a unit-step input of a discrete analog recursive filter is similar to that of the continuous prototype filter except that there are discrete steps of the voltage output, as shown in Fig. 3.11, because the filter is discrete analog and the CCD used is longer than one bit. The feedback coefficients determine the step increments and the frequency characteristics of the filter. Therefore, the time response can also be used to determine the values of the filter coefficients.

1. First-order Discrete Analog Filter

Consider a filter with the transfer function as shown below.

$$H(z) = \frac{a_0 + a_1 z^{-1}}{1 + b_1 z^{-1}}$$

The above transfer function is equivalent to the difference equation which follows.

$$V_0(n) = a_0 V_i(n) + a_1 V_i(n-1) - b_1 V_i(n-1)$$

From table III, the coefficients can be easily calculated by the following relationships.

$$b_1 = -\frac{d_3}{d_2} = -\frac{d_4}{d_3} = -\frac{d_5}{d_4} = \ldots$$

$$= -\frac{d_{n+1}}{d_n} \quad , n = 2, 3, 4, \ldots,$$

$$a_0 = d_1 \; ,$$

$$a_1 = d_2 + b_1 a_0 \, .$$

Illustrations of time responses of several discrete analog recursive filters are in section IV.

2. Second-order Discrete Analog Filter

A general form of a second-order discrete analog recursive filter is described by the transfer function below.

$$H(z) = \frac{a_0 + a_1 z^{-1} + a_2 z^{-2}}{1 + b_1 z^{-1} + b_2 z^{-2}} .$$

The corresponding difference equation is

$$V_o(n) = a_0 V_i(n) + a_1 V_i(n-1) + a_2 V_i(n-2)$$
$$-a_2 V_o(n-1) - b_1 V_o(n-2) .$$

Table IV can be used to find the coefficients of the filter in two parts. By letting $a_1 = a_2 = 0$, b_1 and b_2 can be calculated from

$$d_1 = a_0 ,$$

$$d_2 = -b_1 a_0 ,$$

$$d_3 = b_1^2 a_0 - b_2 a_0 ,$$

which yield

$$b_1 = -(d_2/d_1) ,$$

$$b_2 = -(d_3/d_1) + b_1^2 .$$

After the feedback coefficients have been calculated, a_1 and a_2 can be turned on and their values are calculated by the following relationships.

$$a_1 = d_1 + b_1 a_0 ,$$

$$a_2 = d_2 + b_1(a_1 - b_1 a_0) + b_2 a_0 .$$

Examples of time responses of second-order discrete analog recursive filters are in section IV.

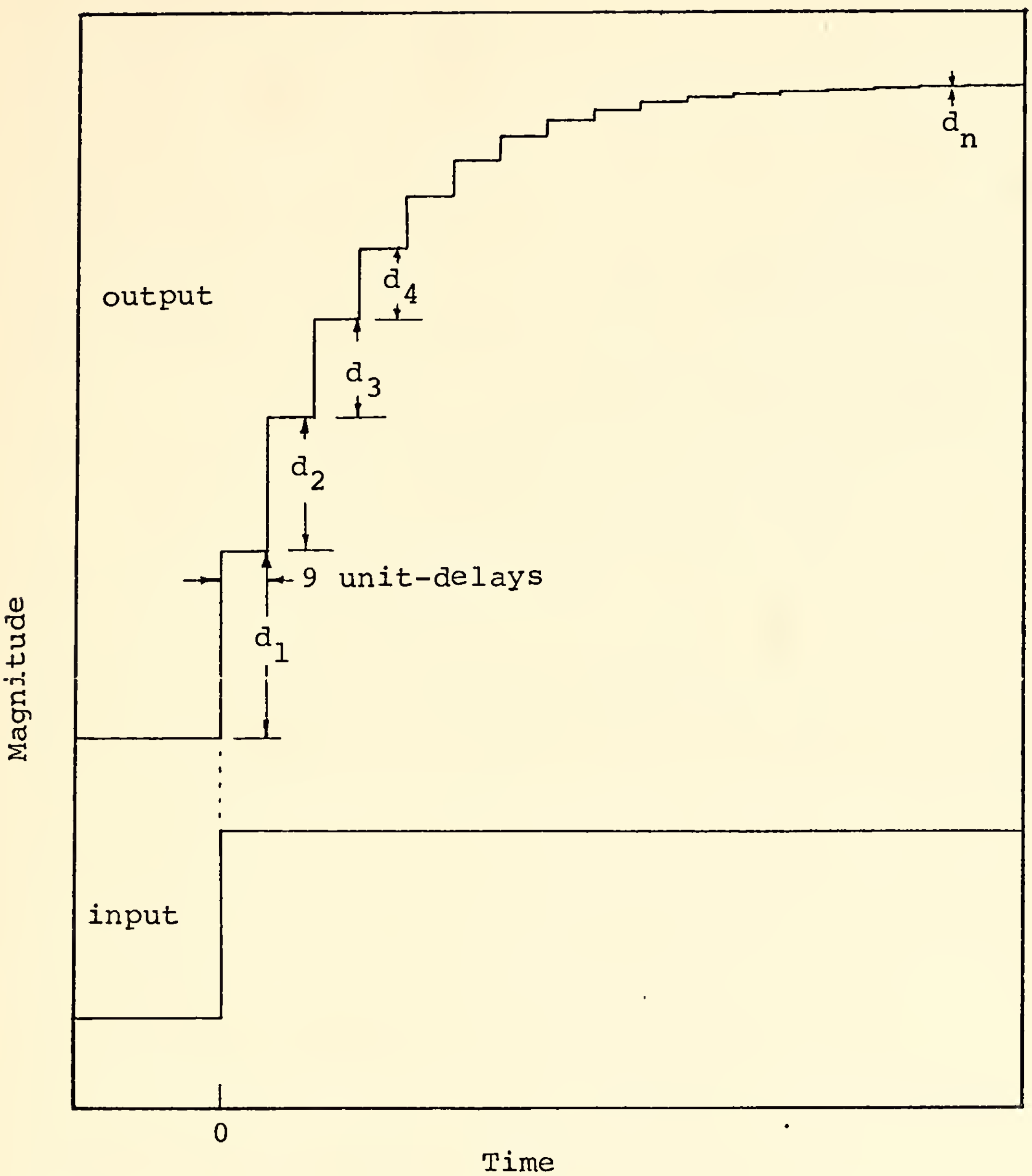


Figure 3.11 Time Response of a Discrete Analog Recursive Filter Using a 9 bit CCD

Table III Time Response to a Unit-step Input of the First-order Discrete Analog Recursive Filter.

n	$V_o(n)$	
0	0	
		$d_1 = a_0$
1	a_0	
		$d_2 = (a_1 - b_1a_0)$
2	$V_0(1) + b_1(a_0 - b_1a_0)$	
		$d_3 = -b_1(a_1 - b_1a_0)$
3	$V_0(2) - b_1(a_1 - b_1a_0)$	
		$d_4 = b_1^2(a_1 - b_1a_0)$
4	$V_0(3) + b_1^2(a_1 - b_1a_0)$	
	etc.	

Table IV Time Response to a Unit-step Input of the Second-order Discrete Analog Recursive Filter.

n	$V_o(n)$	
0	0	
		$d_1 = a_0$
1	a_0	
		$d_2 = a_1 - b_1a_0$
2	$a_0 + a_1 - b_1a_0$	
		$d_3 = a_2 - b_1(a_1 - b_1a_0) - b_2a_0$
3	$V_o(2) + a_2 - b_1(a_1 - b_1a_0) - b_2a_0$	
	etc.	

For both tables above, $d_i = V_0(n+1) - V_0(n)$.

IV. EXPERIMENTAL RESULTS OF DISCRETE ANALOG RECURSIVE FILTER

The objective of the experiments was to investigate the performance of the first-order and the second-order CCD discrete analog recursive filters. At the time, the instrumentation was not adequate for experiments on the second-order CCD filter, therefore, only the first-order CCD filter experiments were conducted. In addition, another type of analog delay, SAD-100 (Serial Analog Delay, manufactured by the Reticon Corporation) was acquired. The SAD-100's have been commercially available and are easy to work with. The experiments on the first-order and the second-order SAD-100 discrete analog recursive filters were conducted. More details of the SAD-100 can be found in IV.B.2.

A. TYPICAL ELECTRICAL OPERATING CONDITION OF THE CCD

At the time of this thesis, there had not been a standardization of how a CCD should be characterized. Many laboratories were working on modelling of the CCD, including a thesis project being carried out at this laboratory.

In order to carry out the experiments on the CCD recursive filtering technique, three sets of curves were obtained to characterize the electrical operation of the CCD. These curves were useful in determining the optimum operating levels of the voltages on the c_1 and c_2 gates. The first set is shown in Fig. 4.1 in which the output a.c. signal gain was plotted against the d.c. voltage on gate c_2, with the voltage on gate c_1 as the varying parameter; a small sinusoidal signal of 20 mv, rms was modulating the voltage on gate c_1. Figure 4.2 is similar to Fig. 4.1 except that the voltage on gate c_2 was being modulated. Each of the two sets of curves mentioned shows the regions in which the small a.c. signal gain was constant. Therefore, the third set of curves shown in Fig. 4.3 was constructed from Fig. 4.1 and 4.2 to show the region of

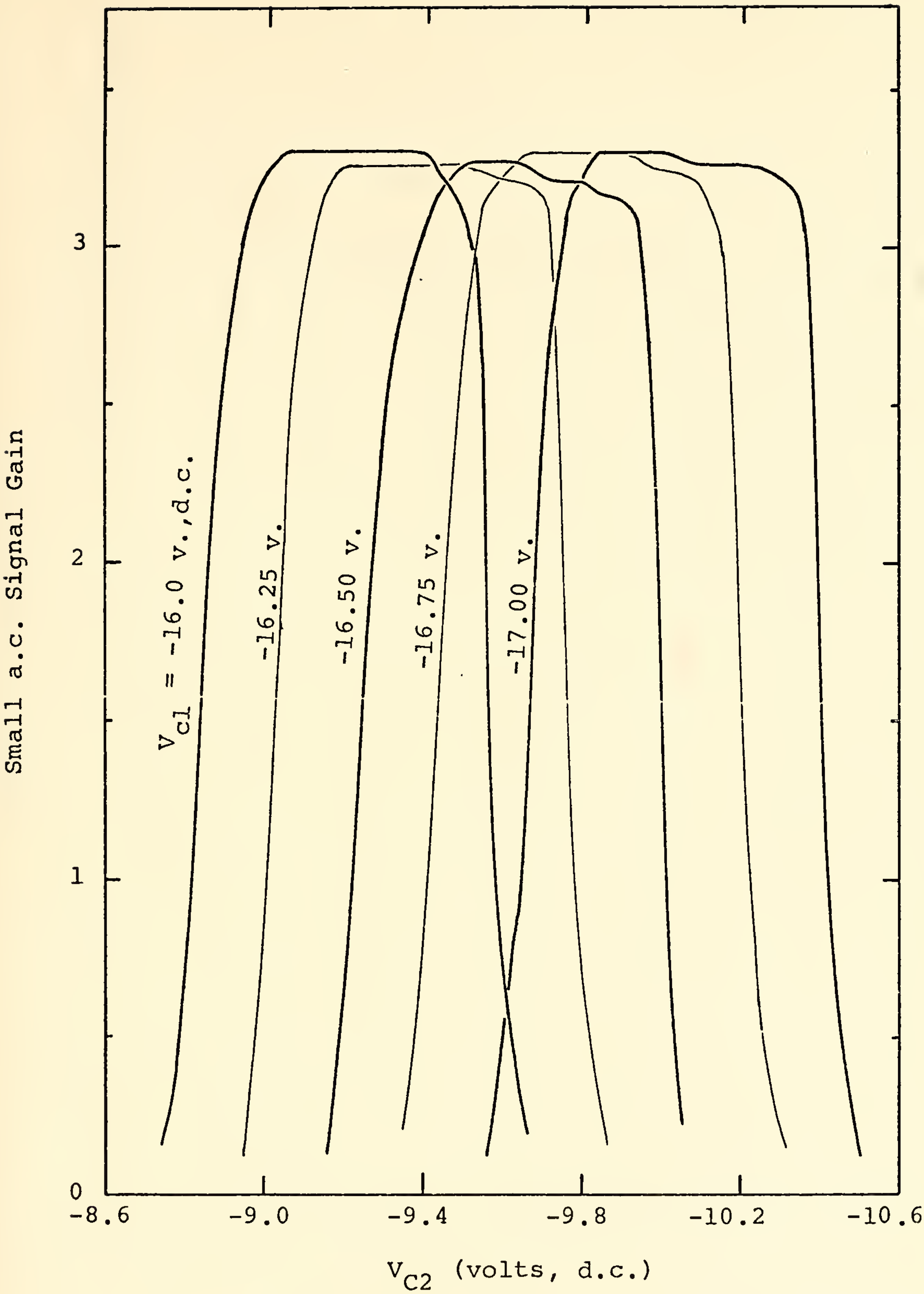


Figure 4.1 CCD a.c. Signal Gain, input on c_1

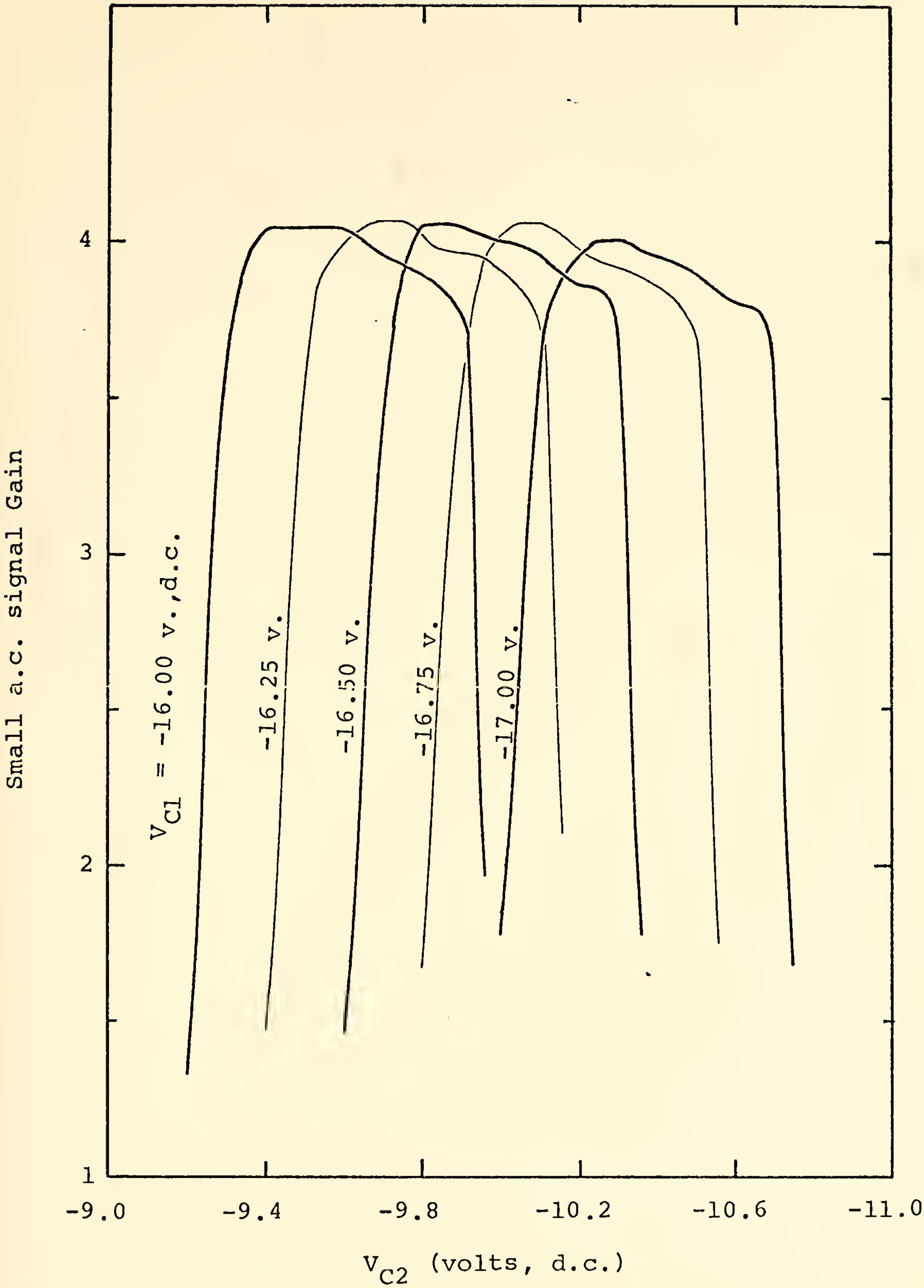


Figure 4.2 CCD a.c. Signal Gain, Input on c_2

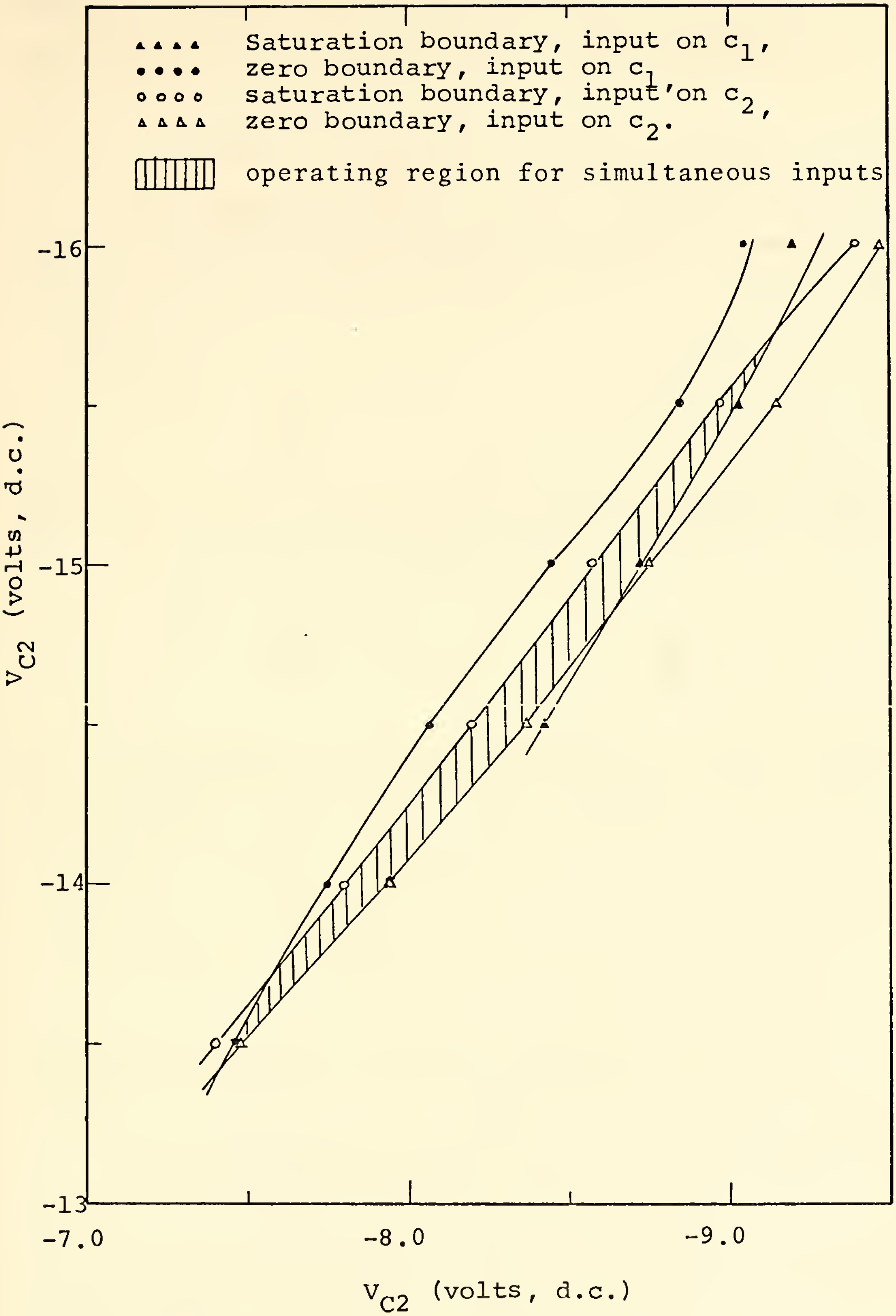


Figure 4.3 Operating Region for Simultaneous Inputs

voltages on gate c_1 and gate c_2 in which the a.c. signal fed into either c_1 or c_2 will be operated with a constant a.c. gain, i.e. in a linear region.

B. FIRST-ORDER DISCRETE ANALOG RECURSIVE FILTER

A simple continuous low-pass prototype filter with cut-off radian frequency at ω_L radian/second was used. Its transfer function in the s-domain is

$$H(s) = \frac{1}{\frac{s}{\omega_L} + 1} . \tag{4.1}$$

Using the frequency transformation, the transfer function of the high-pass filter prototype is

$$H(s) = \frac{1}{\frac{\omega_H}{s} + 1} , \tag{4.2}$$

and the corresponding transfer function in the z-domain was found by using table II. for standard z-transform and bilinear z-transform. Therefore,

$$H(z) = \frac{a_0 + a_1 z^{-1}}{1 + b_1 z^{-1}} . \tag{4.3}$$

For low-pass filter and using standard z-transform

$$a_0 = 1,\ a_1 = 0,\ \text{and}\ b_1 = -e^{-\omega_L T} .$$

For low-pass filter and using bilinear z-transform

$$a_0 = a_1 = 1, \text{ and } b_1 = - \frac{2 - \omega_p T}{2 + \omega_p T}.$$

Note that ω_p is the cut-off radian frequency of the prototype filter and used in calculations only. The actual cut-off frequency of the filter is ω_z which is described by equation 3.4.

For high-pass filter and using bilinear z-transform, it is obtained that $a_1 = -1$ with other coefficients being the same as those of the low-pass filter.

The coefficients were implemented by 20 kiloohms potentiometers.

1. First-order CCD Recursive Filter

A schematic diagram of the CCD chip used in the experiments is shown in Fig. 2.1. Two input techniques were used which are described in section II.A.3. In one case, only one gate, c_1 or c_2, was used together with an analog summer, as shown in Fig. 4.4. In the second case, both gates were used simultaneously as shown in Fig. 4.5. The second method will be very attractive since an arithmatic operation is performed on the CCD. Both input schemes have been successfully demonstrated.

When there is no input, the difference between the surface potential existed under the c_1 and c_2 gates represents the fat zero level. Sinusoidal test signals were used. The fat zero was approximately fifty percent of full potential well. A sample-and hold circuit, Analog Device SHA-2A, was used before feed back to convert the CCD output pulse trains to a continuous signal. In the configuration used, the source follower output was at a negative dc level, and so was the output of the sample-and-hold circuit. Therefore, for the proper recursive operation, a level-shifting circuit was utilized to bring the reference level of the output signal to zero before being fed back. When both input gates were used as shown in Fig. 4.5, two others level-shifters were also used at the c_1 and c_2 gates not only to control the fat zero level, but also to perform summing of the input signal and the feedback signal. For the set-up in Fig. 4.4, a summer with level shifting capability and one other level-shifter were used for the same purpose. A HP-302A wave analyzer was used to measure the frequency response characteristics.

a. Low-pass Filter by Standard Z-Transform

Three experiments were conducted and the results are shown in Fig. 4.6. through Fig. 4.8 and Fig. 4.10.

In the first experiment, a low-pass filter was implemented by using the configuration shown in Fig. 4.4.

The coefficients in equation 4.3 were

$$a_0 = 1.0,\ a_1 = 0,\ \text{and}\ b_1 = 0.63$$

The theoretical cut-off frequency is 551 Hz. The observed cut-off frequency was 550 Hz.

The second experiment used gate c_1 and gate c_2 for on-chip summing operation. The values of coefficients were

$$a_0 = 1.0,\ a_1 = 0,\ \text{and}\ b_1 = 0.718$$

the calculated cut-off frequency for this case is 390 Hz as compared to 400 Hz that was observed.

In the third experiment, a lower b_1 value of 0.392 was used. The calculated cut-off frequency is 1087 Hz. the observed cut-off frequency was 1250 Hz which was slightly higher due to aliasing.

b. Low-pass Filter by Bilinear Z-Transform

Two experiments identical to the last two of the previous section were conducted to demonstrate the design using bilinear z-transformation. The coefficients used were

$$a_0 = 1.0,\ a_1 = 1.0,\ \text{and}\ b_1 = 0.718,\ 0.392$$

In this case, the calculated cut-off frequencies are 386 Hz and 1015 Hz, respectively. Because of the nonlinear frequency warping, the observed cut-off frequencies did not agree with the predicted values. From equation 3.4, they are:

for $b_1 = 0.718$, $f_z = (1/\pi T)\cdot[\tan^{-1}\{\pi T\cdot 386\}] = 383$ Hz

for $b_1 = 0.392$, $f_z = (1/\pi T)\cdot[\tan^{-1}\{\pi T\cdot 1015\}] = 957$ Hz

c. High-pass Filter by Bilinear Z-Transform

For the high-pass CCD discrete analog recursive filter, the experiment set-up was as shown in Fig. 4.4. The coefficients were

$a_0 = 1.0$, $a_1 = -1.0$
$b_1 = -0.41$

The frequency response of this filter is shown in Fig. 4.9 and the voltage time response to a voltage step input is shown in Fig. 4.11. The calculated cut-off frequency is 627 Hz as compared to 600 Hz which was observed.

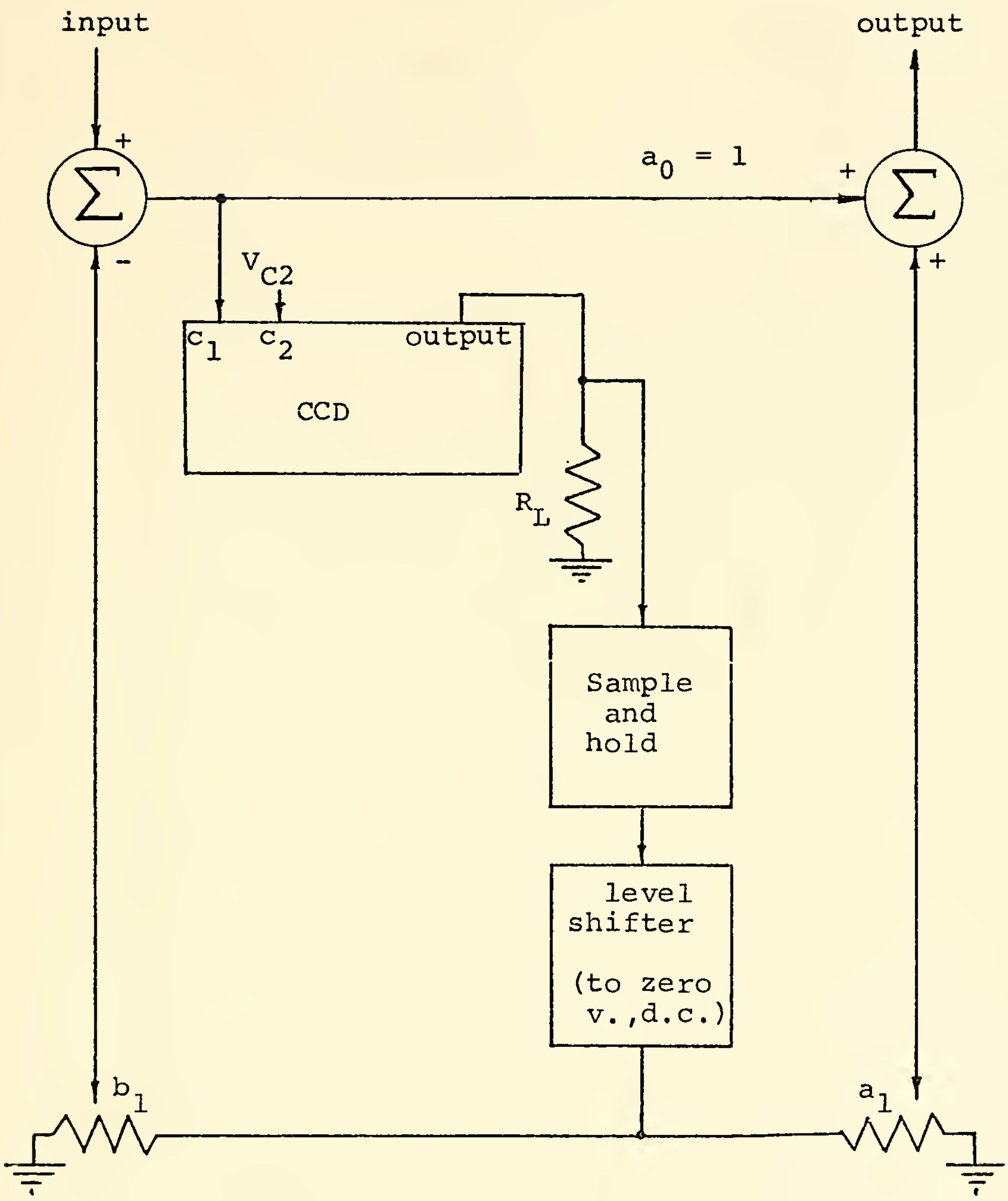


Figure 4.4 Arrangement of the CCD Experiment, Input on c_1 only

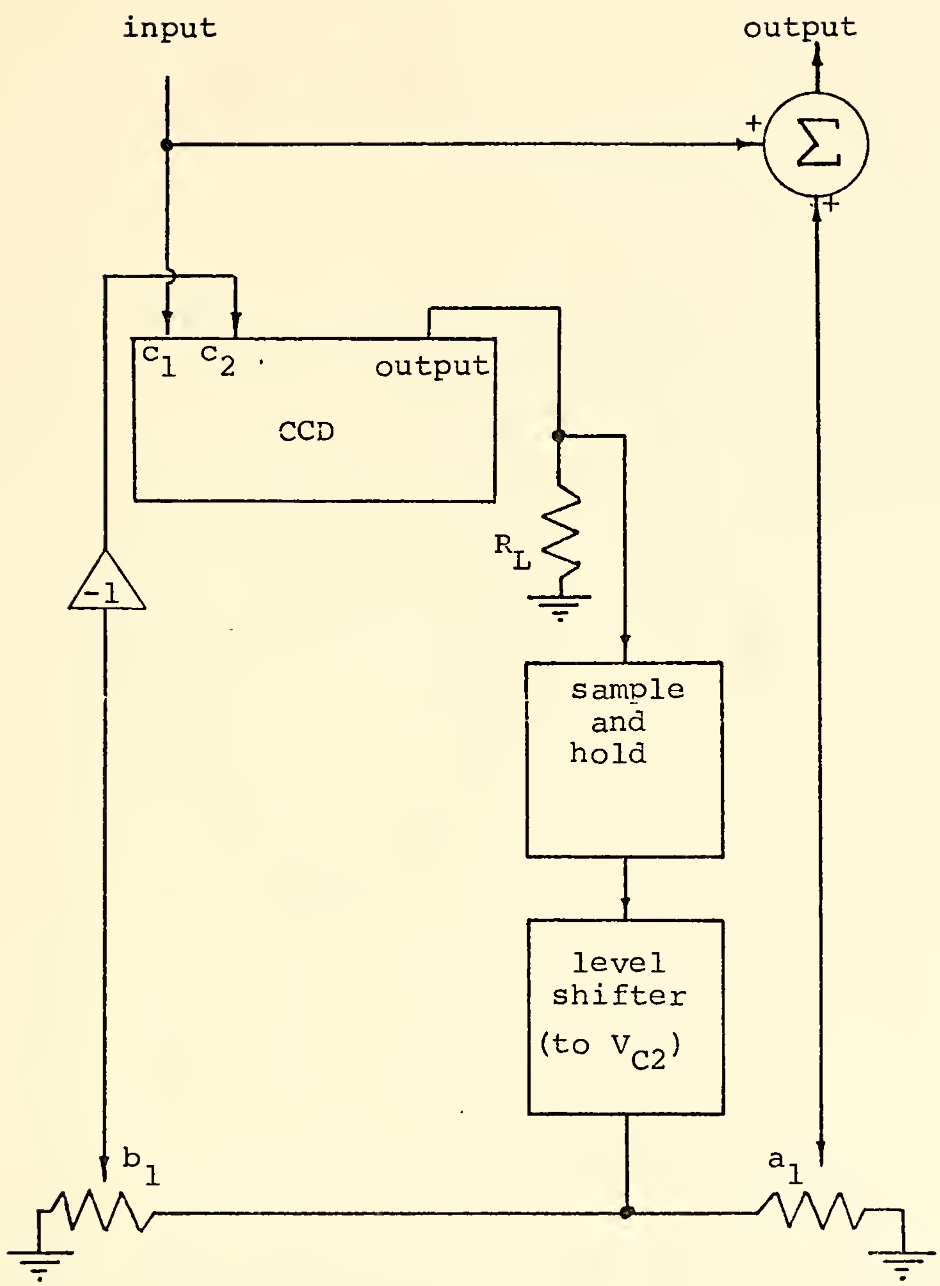


Figure 4.5 Arrangement of the CCD Experiment, Input on c_1 and c_2

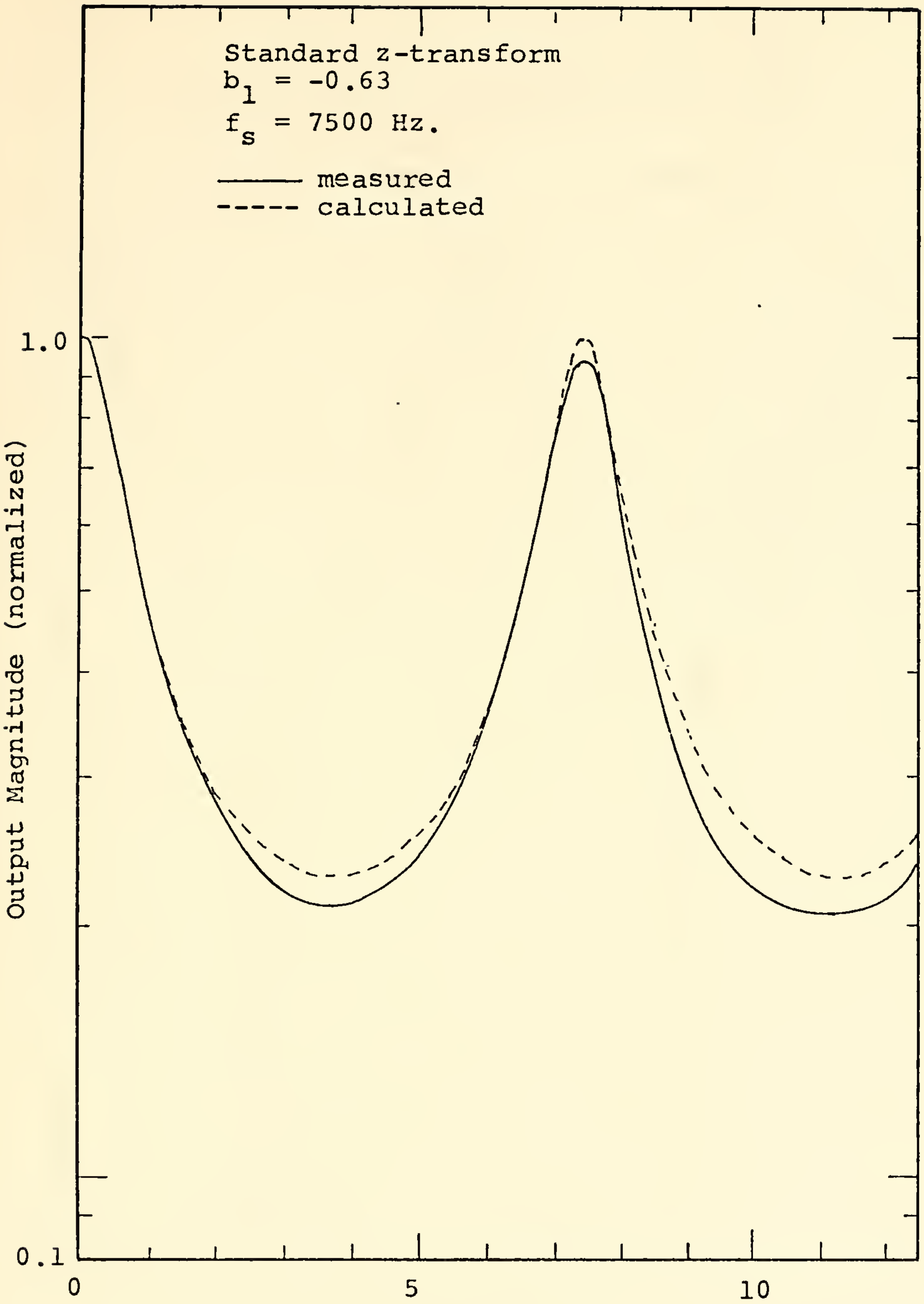


Figure 4.6 Frequency Response of a Low-pass CCD Discrete Analog Recursive Filter, Input on c_1

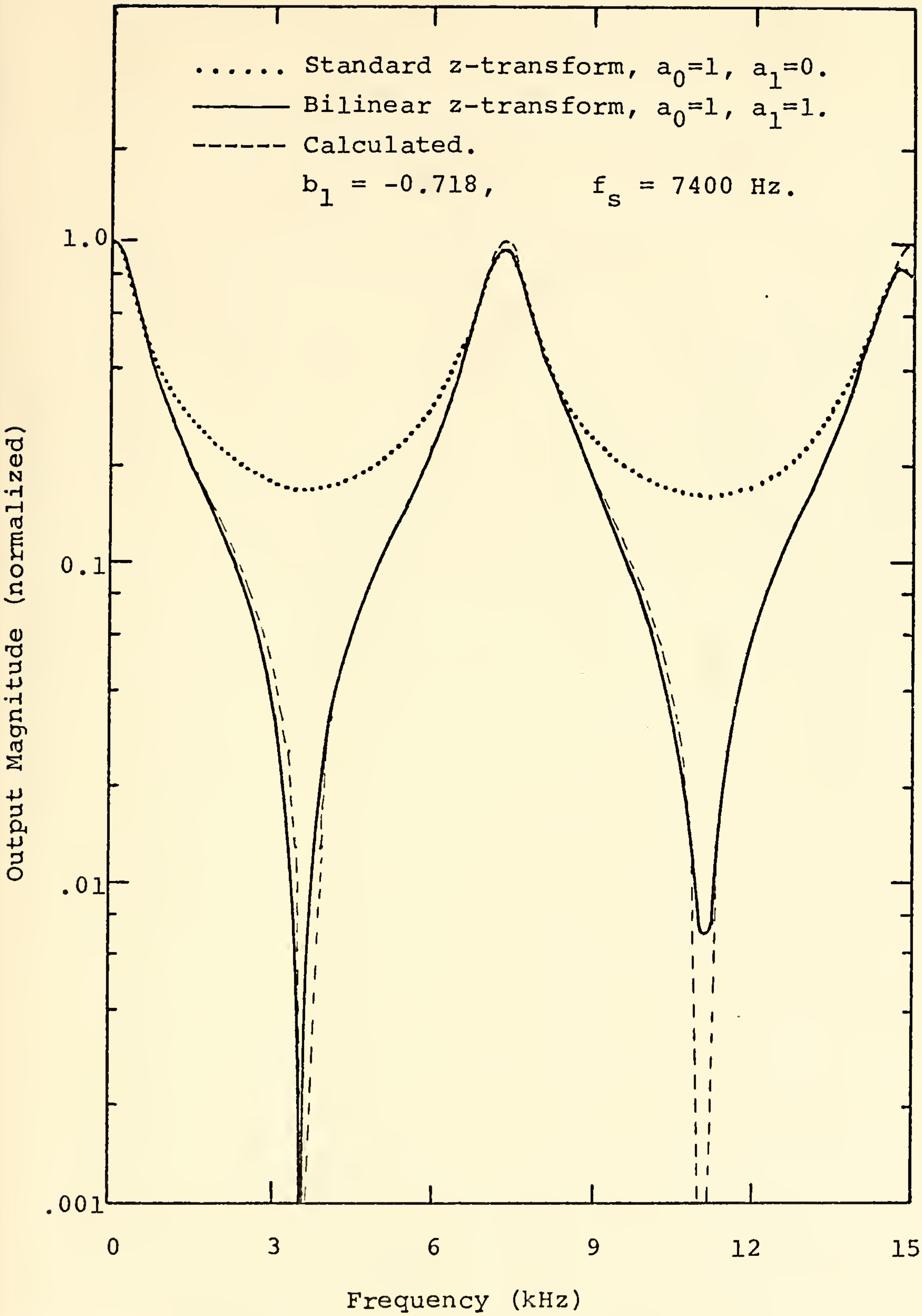


Figure 4.7 Frequency Response of a Low-pass CCD Discrete Analog Recursive Filter, Input on c_1 and c_2

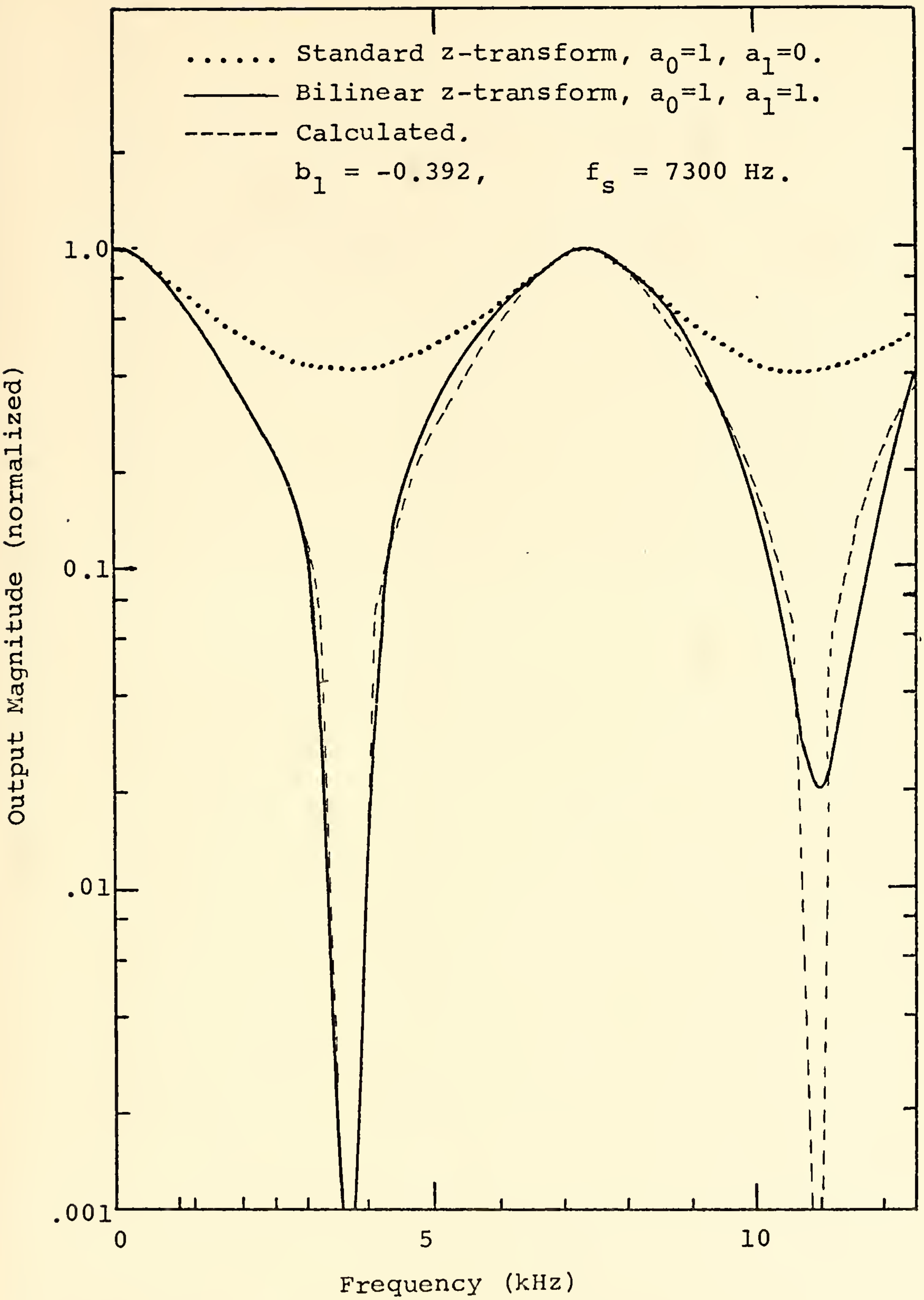


Figure 4.8 Frequency Response of a Low-pass CCD Discrete Analog Recursive Filter, Input on c_1 and c_2

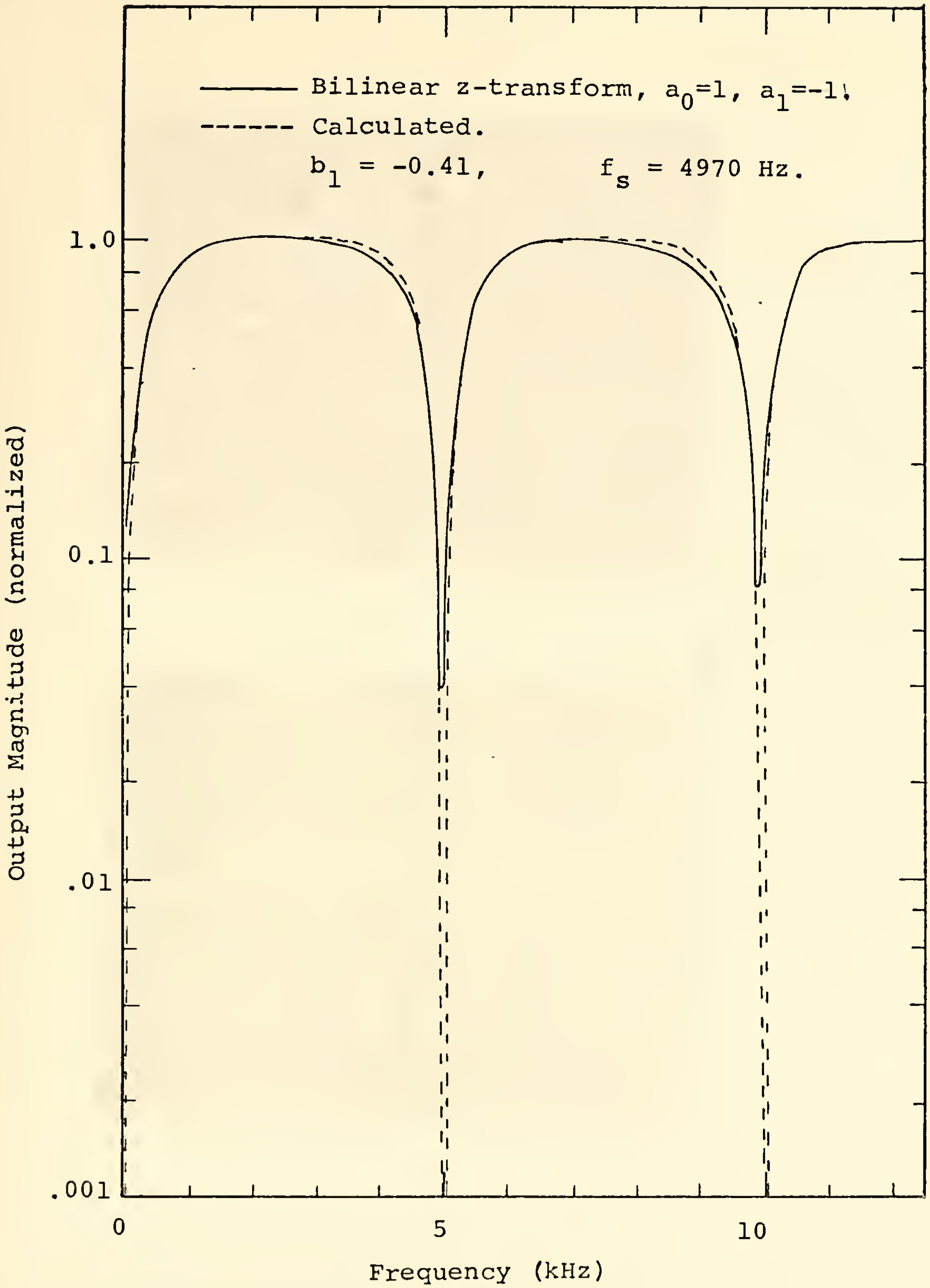
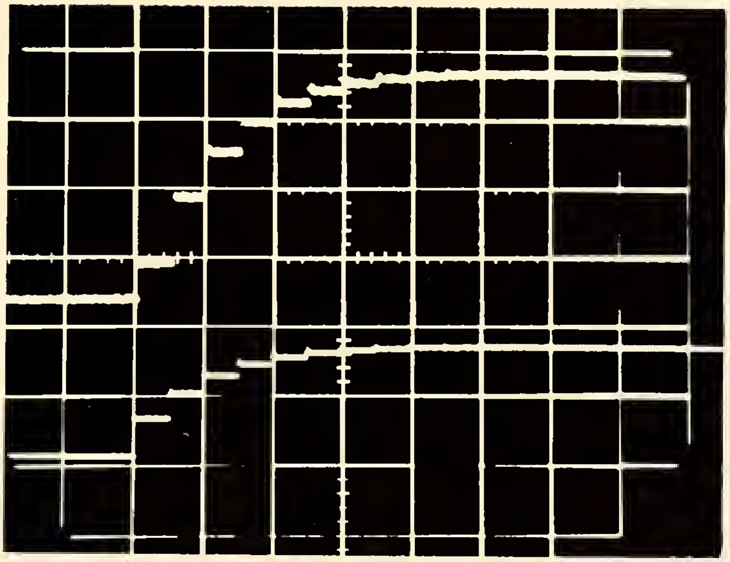


Figure 4.9 Frequency Response of a High-pass CCD Discrete Analog Recursive Filter, Input on c_1.

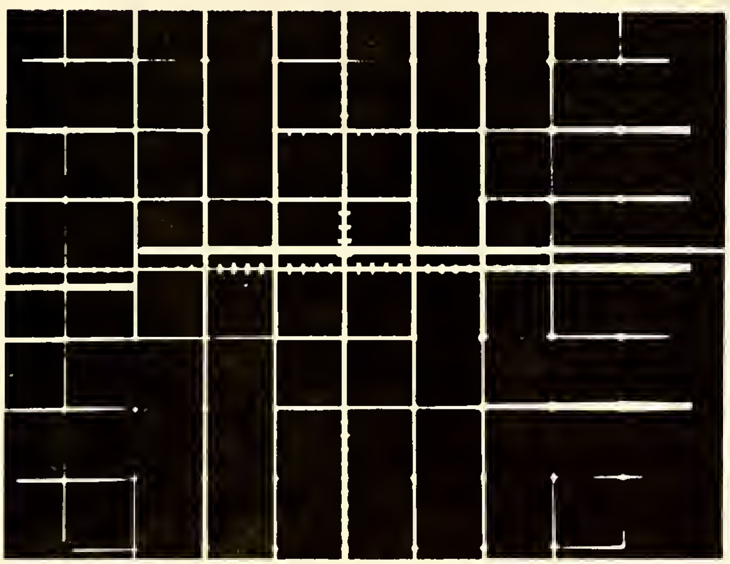
Output



Bilinear
z-transform

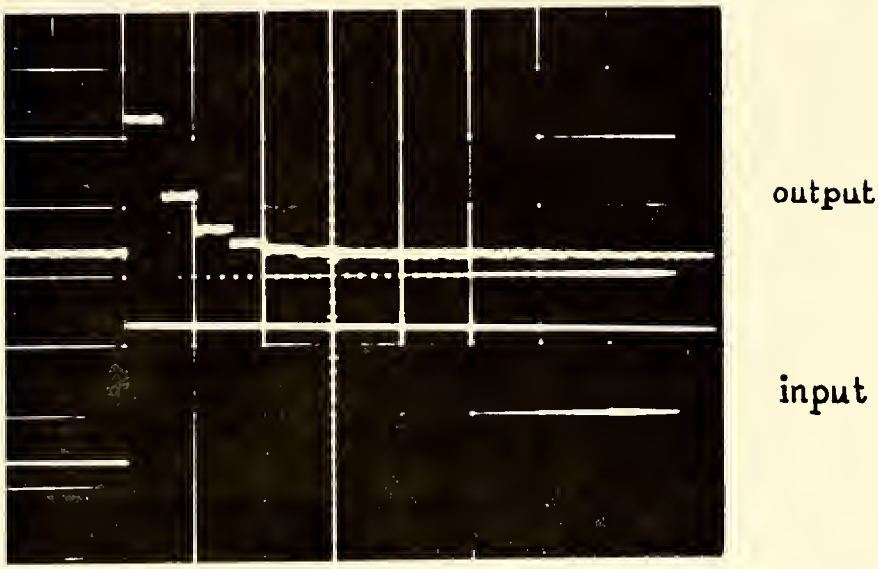
Standard
z-transform

Scales:
vertical: 0.1 v/div.
horizontal:
0.4 msec/div.



Input

Figure 4.10 Time Response to a Step Input of the
Low-pass CCD Discrete Analog Recursive Filter.



Scales: vertical: 0.05 v./div.
horizontal: 0.4 msec/div.

Figure 4.11 Time Response to a Step Input of a High-pass CCD Discrete Analog Recursive Filter.

2. First-order SAD-100 Recursive Filter

The SAD-100 is a monolithic analog memory device which provides temporary storage for analog signal. It has two independent delay lines each with 50 capacitive storage elements operating in parallel to form 100 elements storage device. The storage elements of each delay line are triggered by multiplexed switches which are controlled by a ring counter. Four-phase clocks are used so that the analog information is sequentially sampled in time onto the storage elements. Information is read onto the n-th element while the sample on the (n+1)th element is being read out. The structure of a SAD-100 is shown in Fig. 4.12. Figure 4.13 shows the circuits used for the experiments. The filter results are shown in Fig. 4.14 through Fig. 4.22.

a. Low-pass Filter by Standard Z-Transform

The transfer functions implemented were

$$H(z) = \frac{1}{1 - 0.54z^{-1}}$$

with the calculated cut-off frequency at 2049 Hz. The observed one was 1750 Hz. The result is shown in Fig. 4.14.

and $$H(z) = \frac{1}{1 - 0.20z^{-1}}$$

with the calculated cut-off frequency at 5379 Hz. The observed one was 7000 Hz. The result is shown in Fig. 4.15.

b. Low-pass Filter by Bilinear Z-Transform

The transfer function implemented were

$$H(z) = \frac{1 + z^{-1}}{1 - 0.54z^{-1}}$$

with the calculated cut-off frequency at 1931 Hz. The observed one was 1750 Hz. The result is shown in Fig. 4.14.

and $$H(z) = \frac{1 + z^{-1}}{1 - 0.20z^{-1}}$$

with the calculated cut-off frequency at 3931 Hz. The observed one was 3800 Hz. The result is shown in Fig. 4.15.

c. High-pass Filter by Bilinear Z-Transform

the transfer function implemented was

$$H(z) = \frac{1 - z^{-1}}{1 - 0.20z^{-1}}$$

with the calculated cut-off frequency at 4005 Hz. The observed one was 3700 Hz. The result is shown in Fig. 4.16.

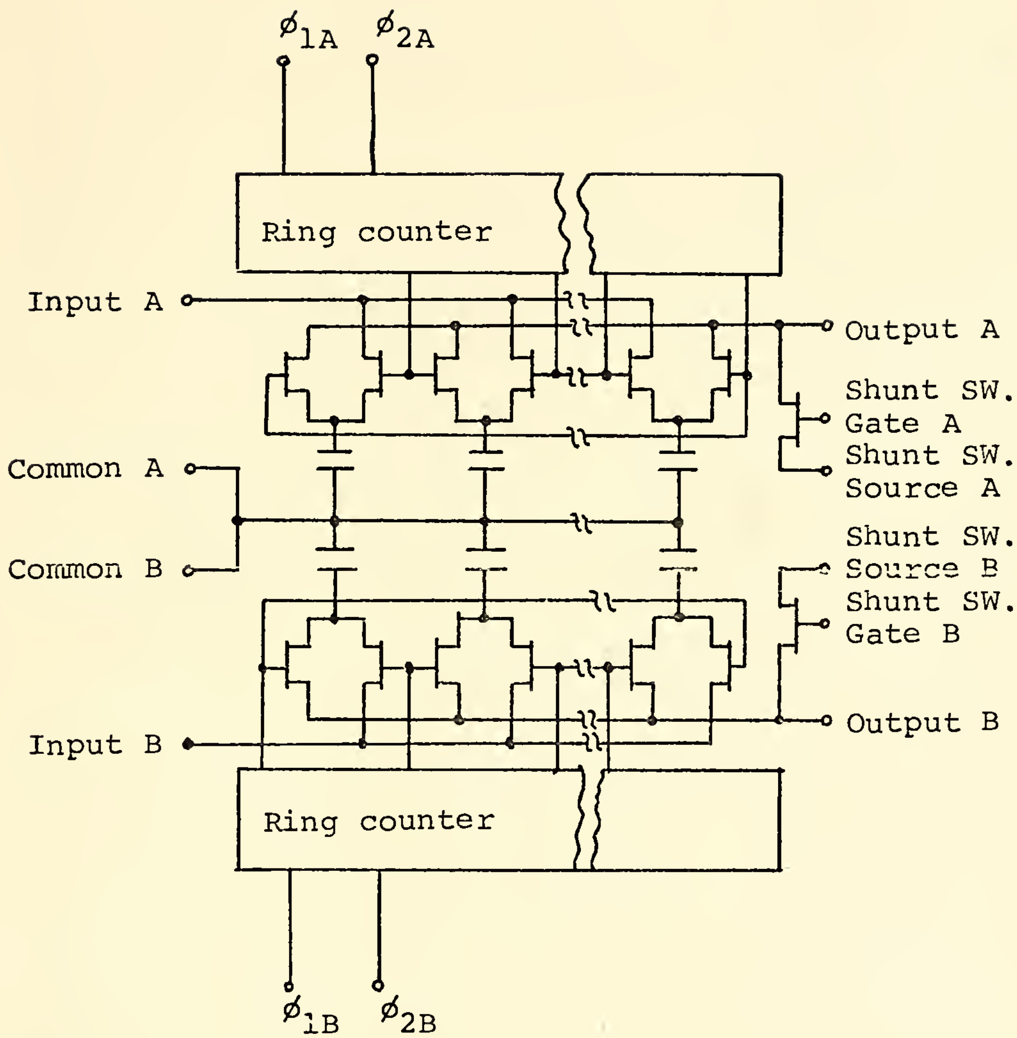


Figure 4.12 Structural Diagram of a SAD-100

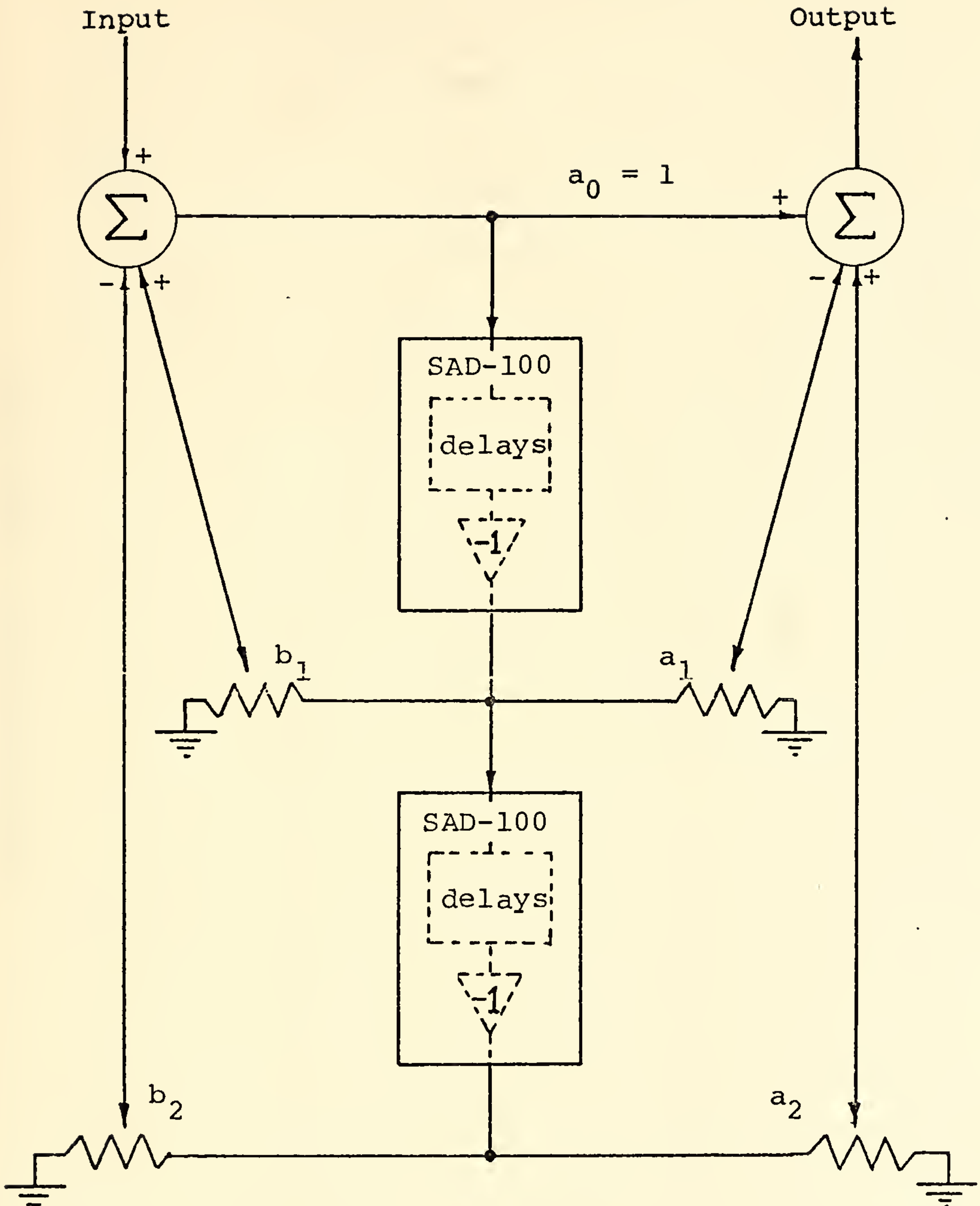


Figure 4.13 Arrangement of SAD-100 Experiments

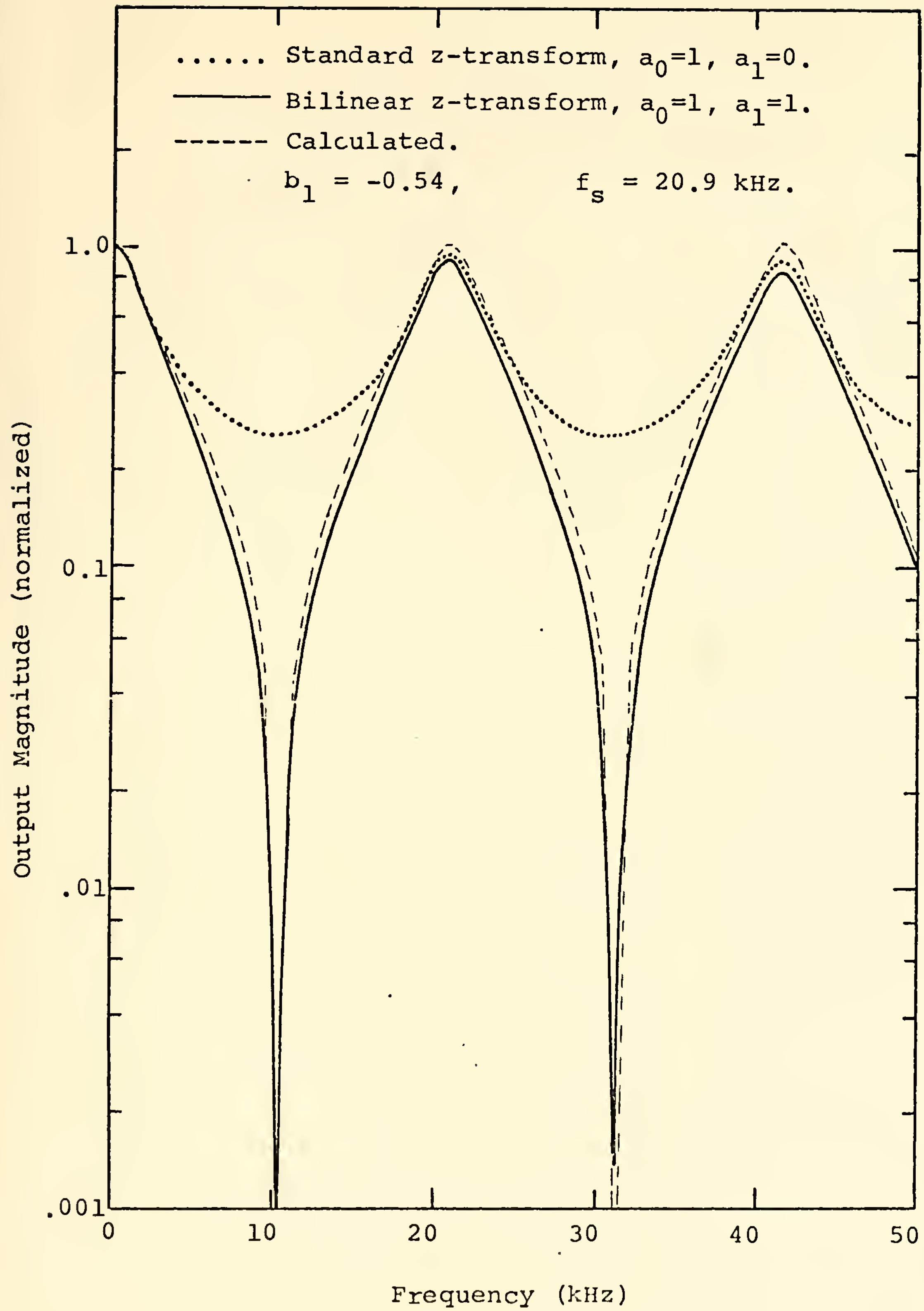


Figure 4.14 Frequency Response of a First-order SAD-100 Low-pass Discrete Analog Recursive Filter

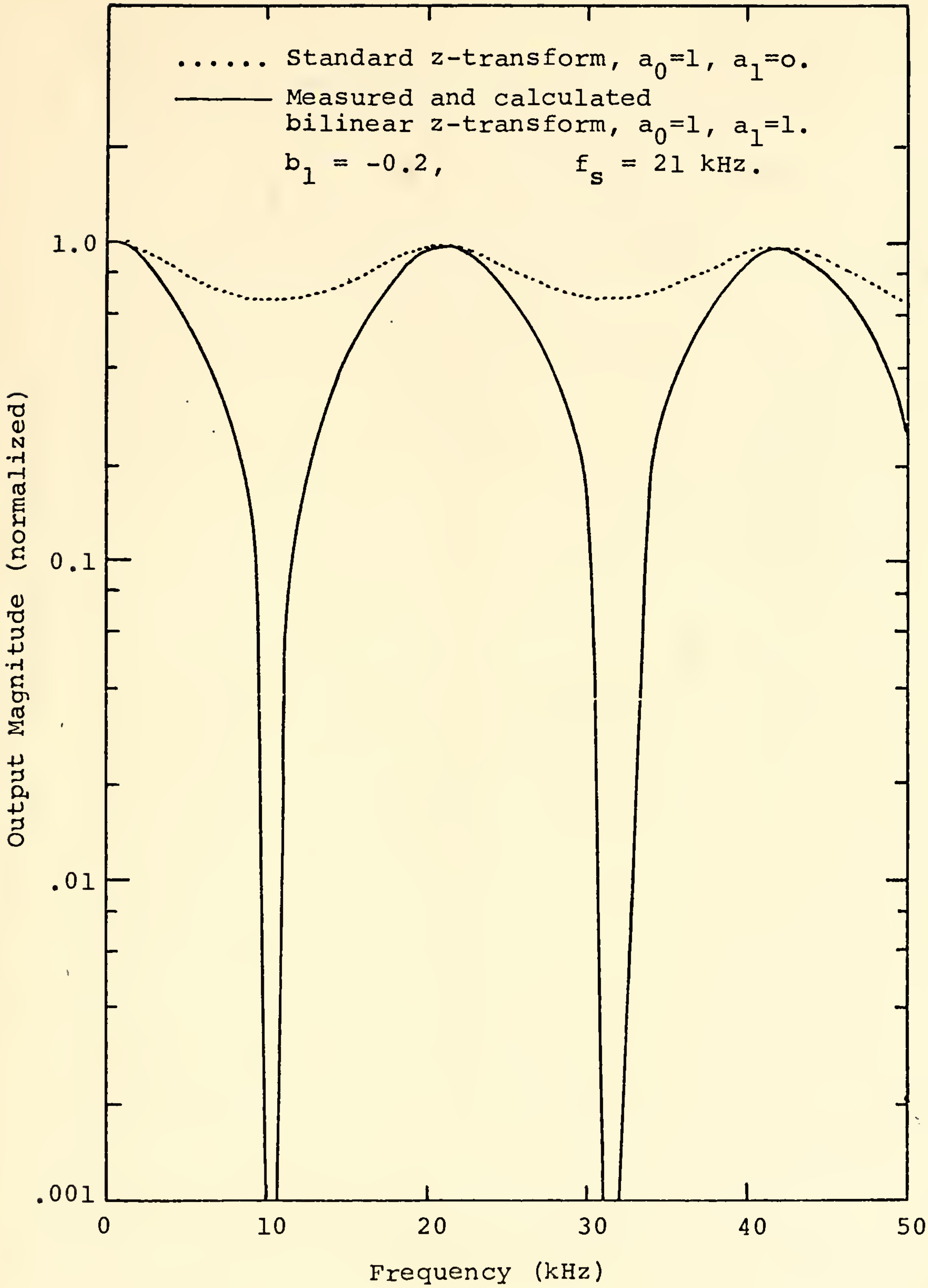


Figure 4.15 Frequency Response of a First-order SAD-100 Low-pass Discrete analog Recursive Filter

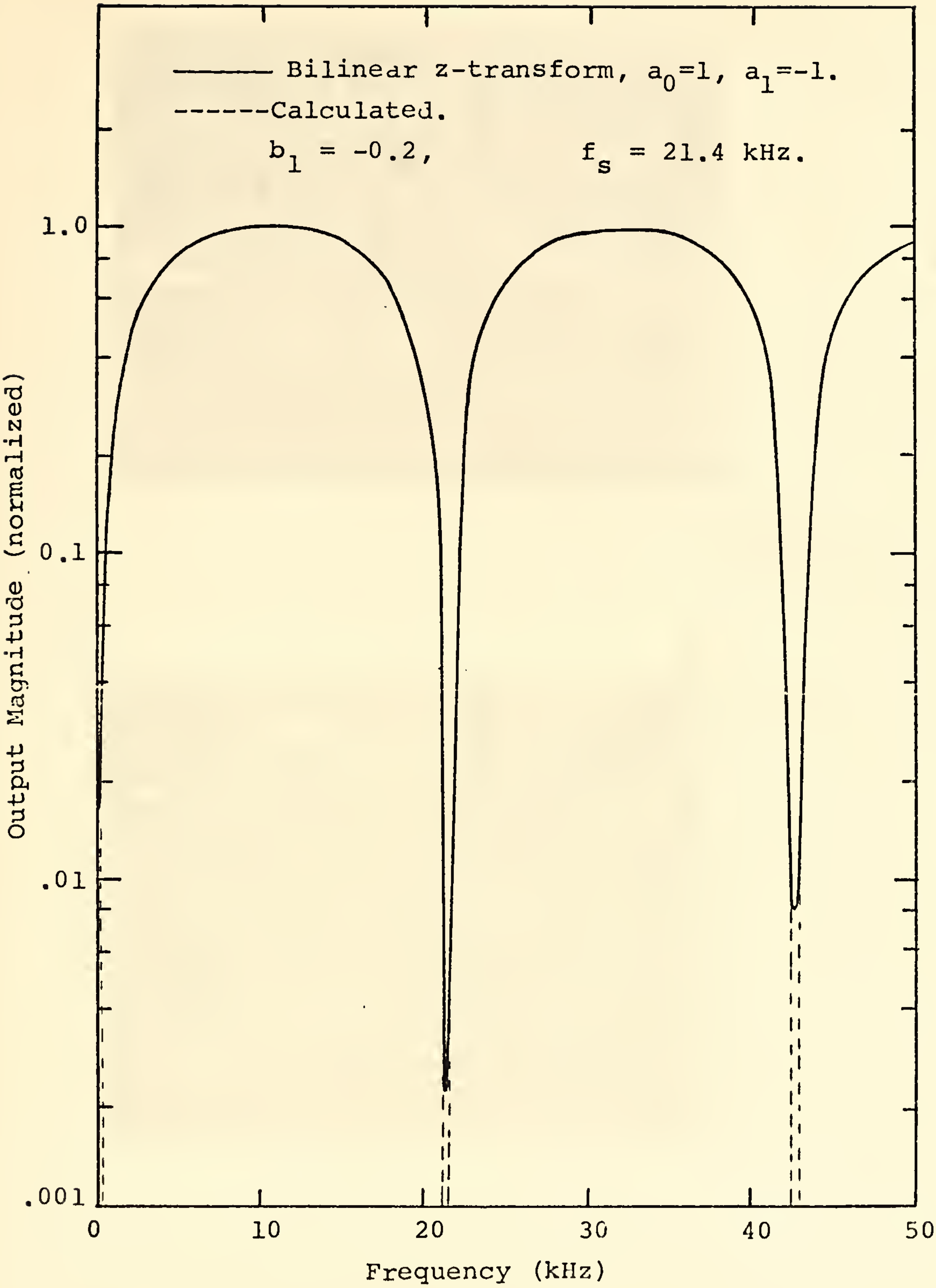
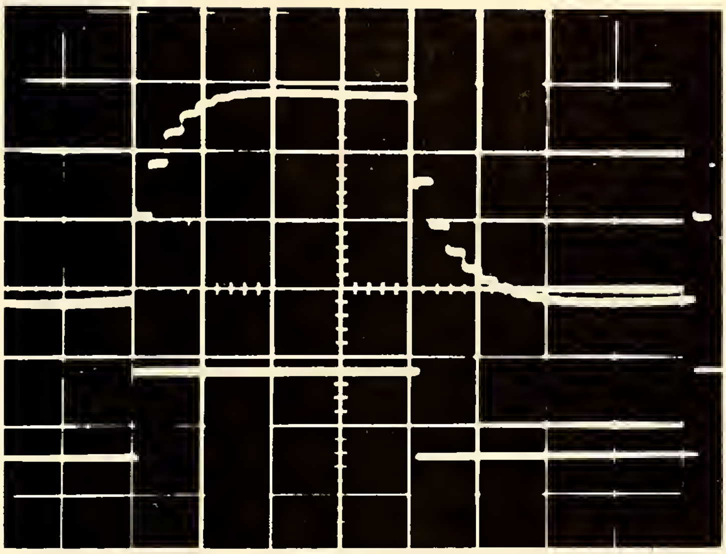


Figure 4.16 Frequency Response of a First-order SAD-100 High-pass Discrete Analog Recursive Filter

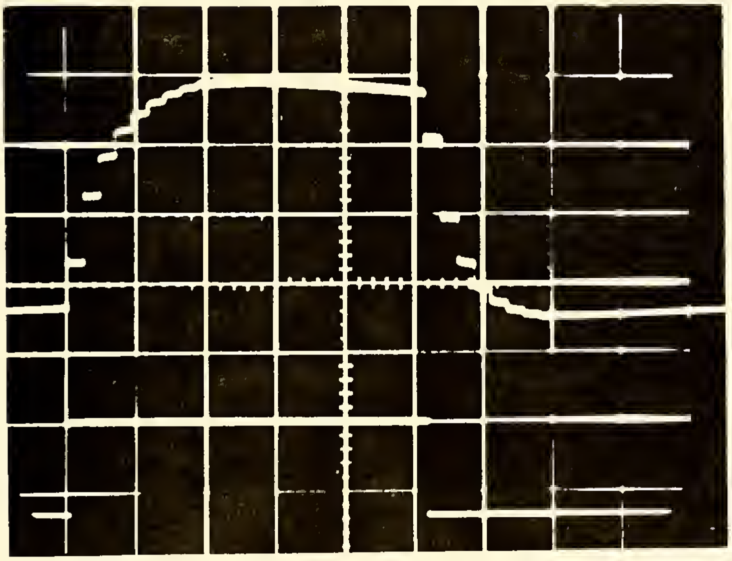


Standard
z-transform

Output

Input

Scales:
vertical:2v./div.
horizontal:
0.2 msec/div.

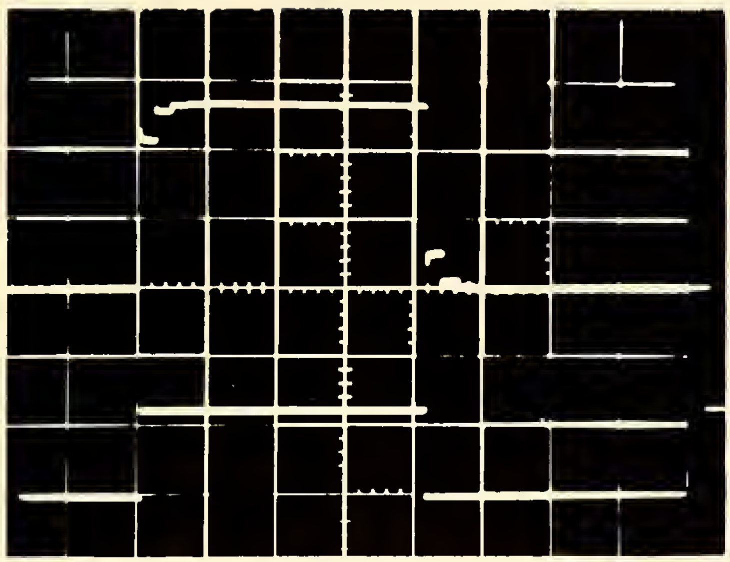


Bilinear
z-transform

Output

Input

Figure 4.17 Time Response to a Step-input of First-order SAD-100 Low-pass Discrete Analog Recursive Filters



Output

Input

Standard z-transform

Scales:

output,	vertical:	1 v./div.
	horizontal:	0.2 msec./div.
input,	vertical:	2 v./div.
	horizontal:	0.2 msec./div.

Figure 4.18 Time Response to a Step-input of a First-order SAD-100 Low-pass Discrete Analog Recursive Filter

C. SECOND-ORDER DISCRETE ANALOG RECURSIVE FILTER

As previously explained, the building block approach would increase the potential of discrete analog recursive filter. Moreover. a second-order block would be more attractive than a first-order one, because it will be possible to implement imaginary coefficients or to obtain a first-order filter by simply setting the second-order coefficients of equation 3.9 to zero.

Only the second-order SAD-100 discrete analog recursive filter experiments were carried out since the instrumentation was not adequate for the second-order CCD filter experiments. The similarities in performances of the first-order discrete analog recursive filters using both types of the devices show that the results of this set of experiments can be used to predict the performances of the second-order filters implemented with the CCD's.

The operating conditions were the same as those used for the first-order SAD-100 filter experiments, i.e. f_s was approximately 21 kHz.

1. Low-pass Filter by Bilinear Z-Transform

A low-pass Butterworth filter was used as the prototype and the cut-off frequency was designed to be 5000 Hz. The coefficients were obtained from table I

$$a_0 = 1, \ a_1 = 1, \ a_2 = 1$$

$$b_1 = -0.281, \ b_2 = 0.186$$

For f_L of 5000 Hz, nonlinear frequency-warping gave the cut-off frequency at 4238 Hz. The observed one was 4200 Hz. The result is shown in Fig. 4.19.

2. High-pass Filter by Bilinear Z-Transform

The same prototype analog filter used for the low-pass filter was also used in this case. The frequency

transformation was carried out and the following coefficients were obtained.

$$a_0 = 1,\ a_1 = -2,\ a_2 = 1$$

$$b_1 = -0.281,\ b_2 = 0.186$$

This resulted in a high-pass filter with the calculated cut-off frequency at 4036 Hz. The result is shown in Fig. 4.20.

3. Band-pass Filter by Bilinear Z-Transform

The prototype analog filter used for this experiment was obtained by cascading two first-order filters, a low-pass and a high-pass. The passband of this filter has calculated 3 db corner frequencies at 1538 Hz and 6355 Hz. The result is shown in Fig. 4.21 and Fig. 4.22.

$$H(z) = \frac{1 + z^{-1}}{1 - 0.10z^{-1}} \bullet \frac{1 - z^{-1}}{1 - 0.40z^{-1}}$$

$$= \frac{1 - z^{-2}}{1 - 0.5z^{-1} + 0.04z^{-2}}$$

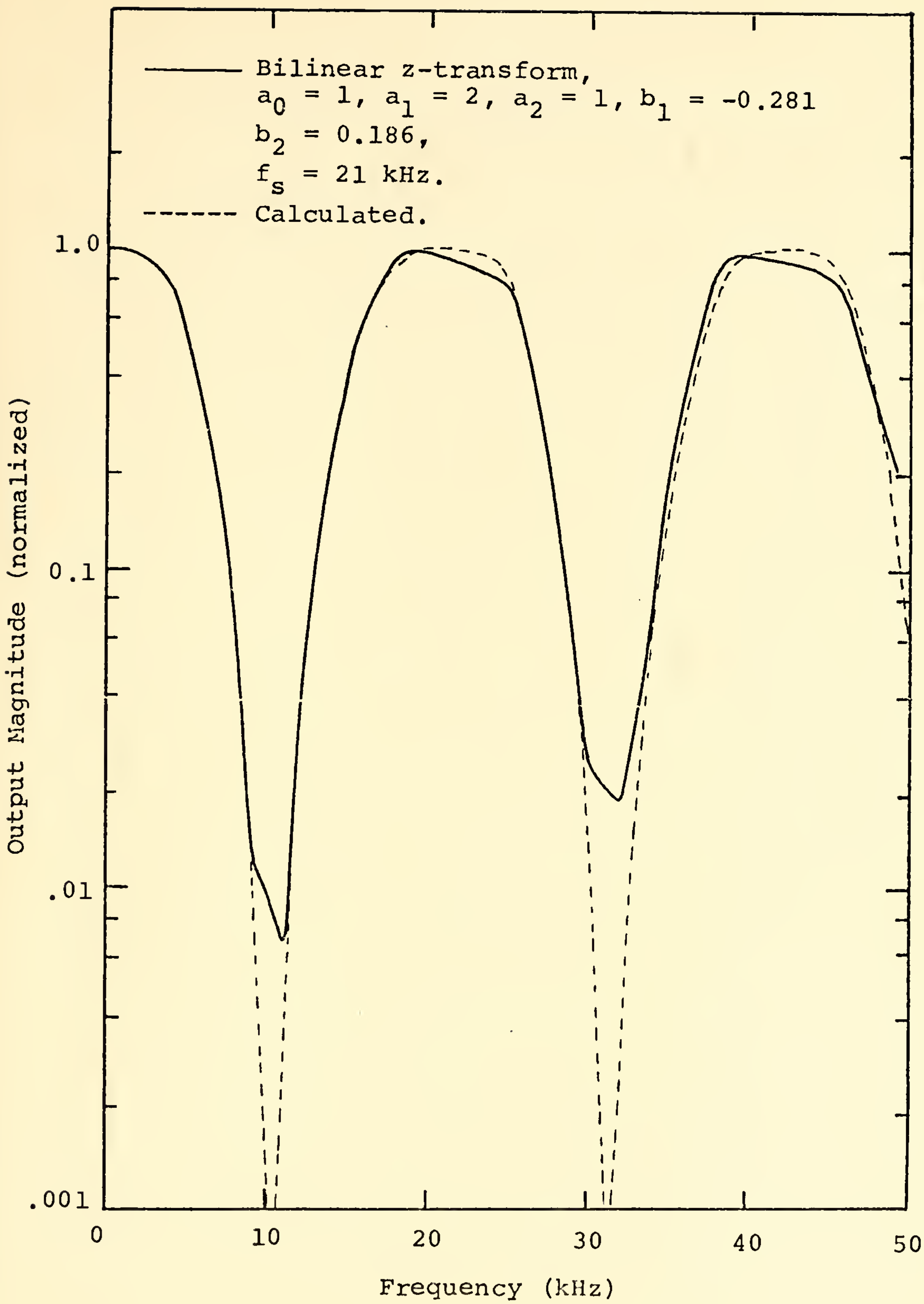


Figure 4.19 Frequency Response of a Second-order SAD-100 Low-pass Discrete Analog Recursive Filter

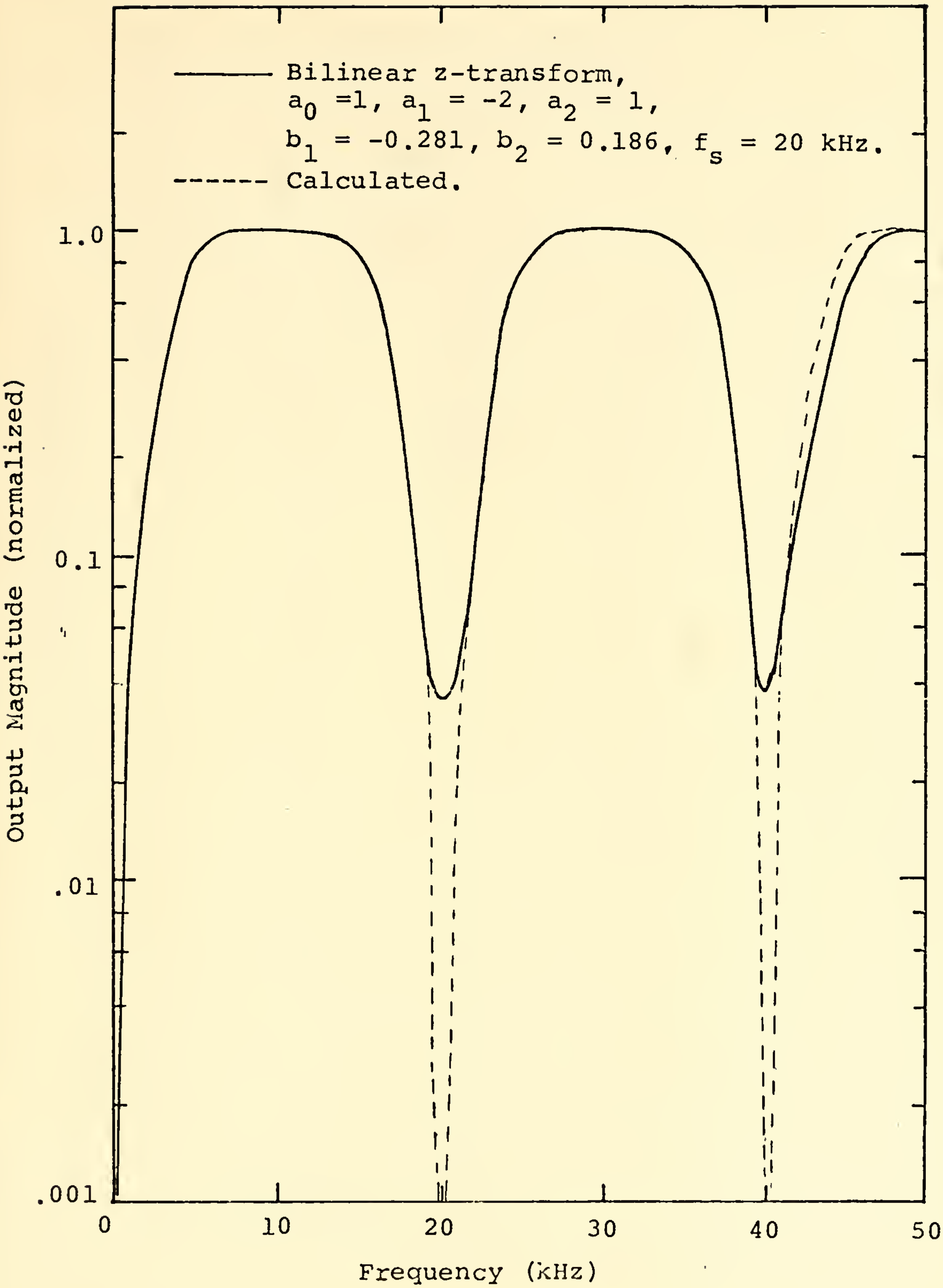


Figure 4.20 Frequency Response of a Second-order SAD-100 High-pass Discrete Analog Recursive Filter

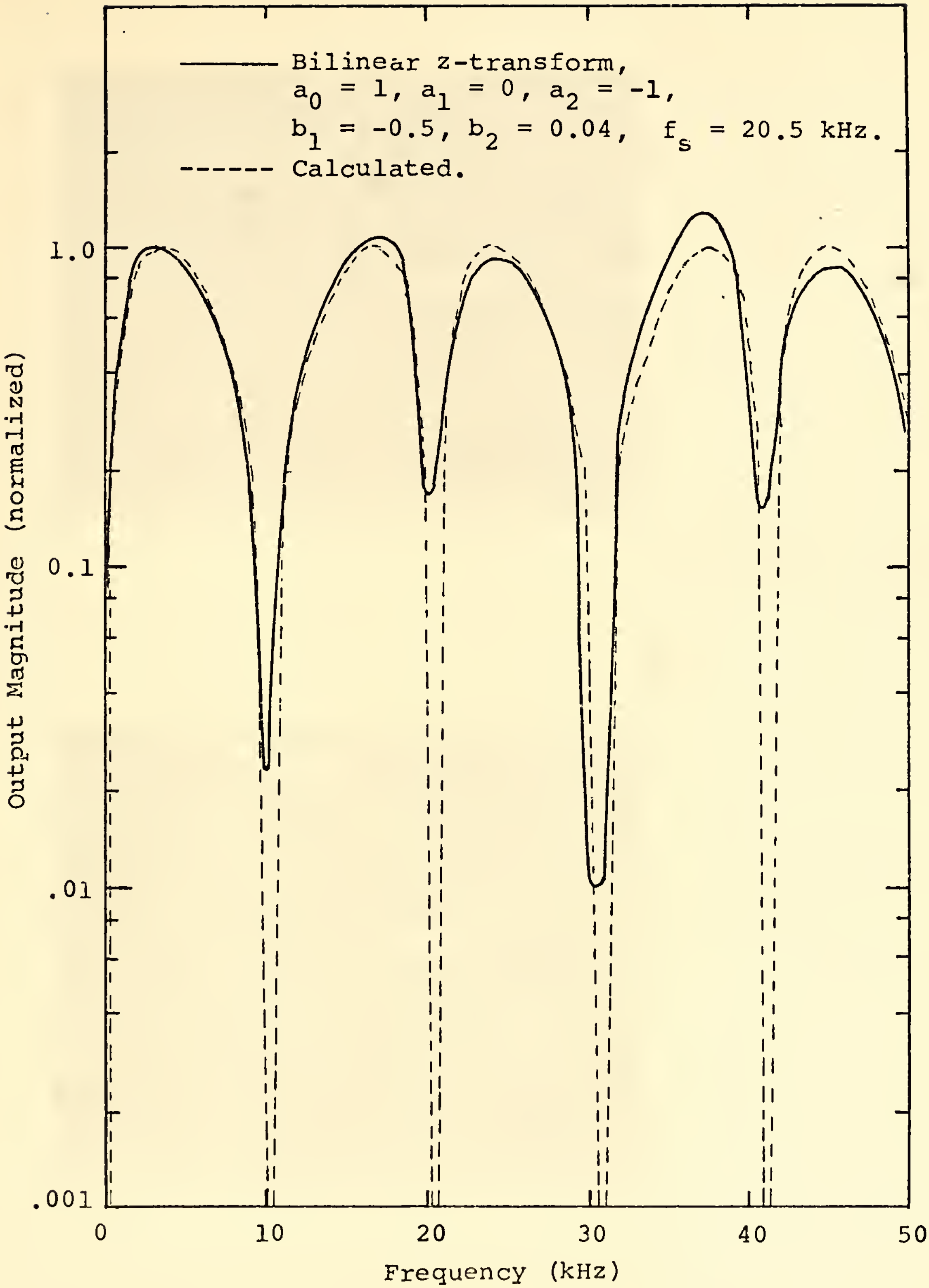
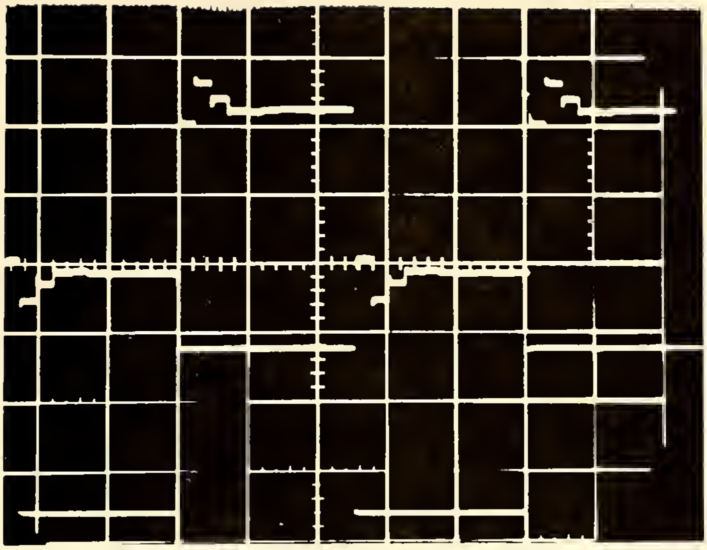


Figure 4.21 Frequency Response of a Second-order SAD-100 Band-pass Discrete Analog Recursive Filter

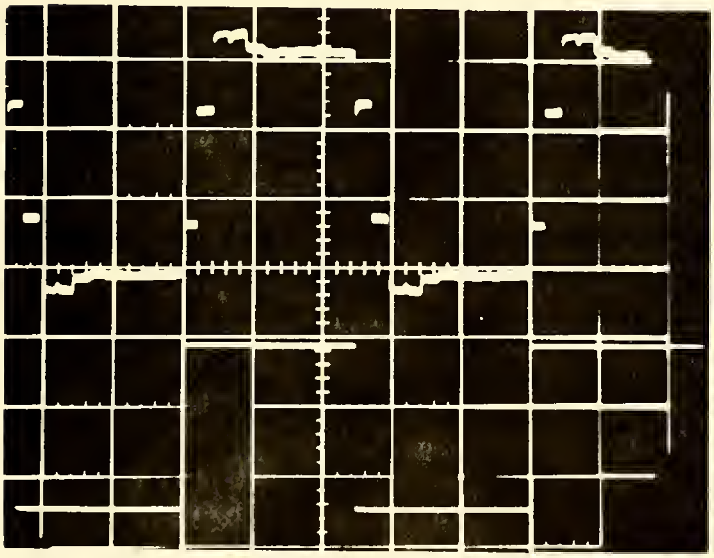


Output

$a_1 = a_2 = 0,$
$a_0 = 1,$
to determine b's

Input

Scales:
vertical: 1 v./div.
hor.: 0.2 msec/div.



Output

$a_o = 1,\ a_2 = -1,$
$a_1 = 0.$
$b_1 = -0.281,\ b_2 = 0.186.$

Input

Figure 4.22 Time Response to a Step-input of a Second-order SAD-100 Discrete Analog Recursive Filter

D. SUMMARY

The experiments on the CCD discrete analog recursive filter were successful. With attention given to the dc level of the feedback signal, filters using CCD performed as expected. The a.c. signal gain of the CCD's used was greater than unity, as a result, a wide range of coefficients, thus a wide range of critical frequencies, was possible. Because a CCD has a finite dynamic range, the coefficients had an upper limit after which the CCD would be saturated. However, this limit was close to unity for a first-order filter and it could still be designed to have practically any critical frequency smaller than one-half of the effective sampling frequency.

Two very important aspects of CCD discrete analog recursive filter should be emphasized. Firstly, the filters designed by the standard z-transform method and the bilinear z-transform method performed differently as can be seen in Fig. 4.7 and Fig. 4.8. The stopband attenuation obtained with the standard z-transform implementation was always less than 20 db and aliasing effect was severe when b_1 was at low values. In contrast, the stopband attenuation obtained with the bilinear z-transform method was always greater than 40 db. It could be much higher with better coefficient accuracy and environmental condition. Therefore, with the bilinear z-transform design, there is really no need to use a high-order filter to obtain great stopband attenuation. Secondly, because the delays used have more than one bit which makes the effective sampling frequency much lower than the clock frequency, the frequency characteristics of these filters have a comb-filter effects as shown in Fig. 4.7 and Fig. 4.8. For the CCD case, the clock frequency was 67.50 kHz, the effective sampling frequency of the filter was 7.5 kHz. For a low-pass CCD filter design, the frequency response was periodic in frequency with a period of 7.5 kHz. The usable frequency range was 33.75 kHz. The result was a

comb-filter with center frequencies at 0 Hz, 15.0 kHz, 22.5 kHz, and 30.0 kHz, and all passbands had equal bandwidth of twice the designed corner frequency.

The experiments of the filters using the SAD-100's were less complicated than those using the CCD's, because the SAD-100 are now better developed and in production. The filter results using these two IC delays support one another. This was helpful in the second-order discrete analog recursive filter case since it was not feasible to perform experiment of the second-order CCD discrete analog filter at the time. If the cut-off rate at the stopband is the main performance criteria, the results showed that the first-order discrete analog recursive filters, having less hardware to deal with, were not inferior to the second-order ones when the bilinear z-transform method was used.

The following tables give the summary of performances of the discrete analog recursive filters implemented.

Table V First-order Low-pass Discrete Analog Recursive Filters

		b_1	f_L, (Hz) calculated	f_L, (Hz) measured	f_S (kHz)
CCD	Standard	0.63	551	550	7.5
	z-transform	0.718	390	400	7.4
		0.392	1087	1250	7.3
	Bilinear	0.718	383	360	7.4
	z-transform	0.392	957	1000	7.3
SAD-100	Standard	0.54	2049	1750	20.9
	z-transform	0.20	5379	7000	21.0
	Bilinear	0.54	1931	1750	20.9
	z-transform	0.20	3931	3800	21.0

Table VI First-order High-pass Discrete Analog Recursive Filter (Bilinear Z-Transform).

	b_1	f_L, (Hz) calculated	f_L, (Hz) measured	f_s (kHz)
CCD	-0.41	4005	3700	4.97
SAD-100	-0.20	627	600	21.4

Table VII Second-order SAD-100 Discrete Analog Recursive Filters (Bilinear Z-Transform)

	a_0	a_1	a_2	b_1	b_2	cut-off frequencies calc.	meas.	f_s (kHz)
low-pass	1	2	1	-0.281	0.186	4238	4200	21.0
high-pass	1	-2	1	-0.281	0.186	4036	4200	20.0
band-pass	1	0	-1	-0.5	0.040	1538	1300	20.5
						6355	6000	

V. DESIGN PROCEDURES OF DISCRETE ANALOG RECURSIVE FILTER

A. DESIGN STEPS

The versatility of the building block approach has been discussed and the following design procedures are based on the second-order building block. The transfer function is shown below again for convenience.

$$H(z) = \frac{a_0 + a_1 z^{-1} + a_2 z^{-2}}{1 + b_1 z^{-1} + b_2 z^{-2}}. \tag{5.1}$$

The design of a discrete analog recursive filter can be divided into four steps as follows.

1. Step 1

From the desired specifications, determine the transfer function of a low-pass continuous filter to be used as the prototype [6]. The resulted transfer function is then transformed into the s-domain by the Laplace transformation and expressed in terms of (s/ω_L).

2. Step 2

If the desired filter is not the low-pass type, use the following frequency transformations.

For a high-pass filter: $\frac{s}{\omega_L} \Rightarrow \frac{\omega_H}{s}$.

For a band-pass filter: $\frac{s}{\omega_L} \Rightarrow \frac{s^2 + \omega_0^2}{s}$.

For a band-stop filter: $\frac{s}{\omega_L} \Rightarrow \frac{s}{s^2 + \omega_0^2}$.

3. Step 3

The transfer function obtained from the previous step is then transformed into the z-domain by appropriate methods described in section III.B.1.

4. Step 4

If the degree of the transfer function, H(s), of the desired discrete analog recursive filter is higher than second-order, it must be factored into a product or a sum of first-order and second-order functions. Each first-order and/or second-order function are first implemented. All the sections are then cascaded or connected in parallel according to the way the original transfer function was factored.

B. COEFFICIENTS SETTING

1. Feedback Coefficients

In order to be able to set the coefficients at desired values, a calibration chart shown in Fig. 5.1 was constructed for the feedback coefficients, b_1 and b_2. The values of each coefficient, implemented by a ten-turn potentiometer, were plotted against the potentiometer readings. The graphs were obtained by using the time response to a step input of a discrete analog recursive filter with the transfer function as shown in equation 5.1, while a_1 and a_2 were set to zero. The plotted values of b_1 and b_2 were determined by procedures outlined in section III.D. Note that a potentiometer setting will give the magnitude of the corresponding coefficient regardless of the sign, which is separately set at the summer. For a first-order discrete analog recursive filter, b_2 is simply set to zero, i.e. open-circuited.

2. Feed-forward Coefficients

Since a transfer function can be changed by a constant factor, it is necessary only to eatablish the correct relative values of the feed-forward coefficients, a_0, a_1, and a_2. For a second-order discrete analog recursive filter, it is obvious from section III.D.2 that if $b_1 = b_2 = 0$, then the time response to a step input gives

$$a_0 = d_1,$$

$$a_1 = d_2,$$

$$a_2 = d_3.$$

Therefore, the coeficient values can be set by using an oscilloscope.

For the bilinear z-transform, the accuracy of these feed-forward coefficients can be easily checked since the relationship among them is always fixed. For the first-order one, a_0 is always equal to a_1; and for the second-order, $2a_0 = 2a_2 = a_1$. It was known from the previous discussion that the transfer function of a first-order low-pass discrete analog recursive filter is

$$H(z) = \frac{1 + z^{-1}}{1 - b_1 z^{-1}},$$

which produces a zero at z = −1, or zero-magnitude transfer at $f_c/2$ Hz . Therefore, in the second-order building block implementation shown in Fig. 3.9, setting b_1 and a_1 to zeroes and b_2 to some finite value also results in a low-pass filter. A sinusoidal input signal is then fed into the input port. When a_0 and a_2 are adjusted to give zero output, a_0 will be equal to a_2. The second-order coefficients can then be turned on, and when a_1 is adjusted so that the output is again zero, correct setting of all coefficients are assured.

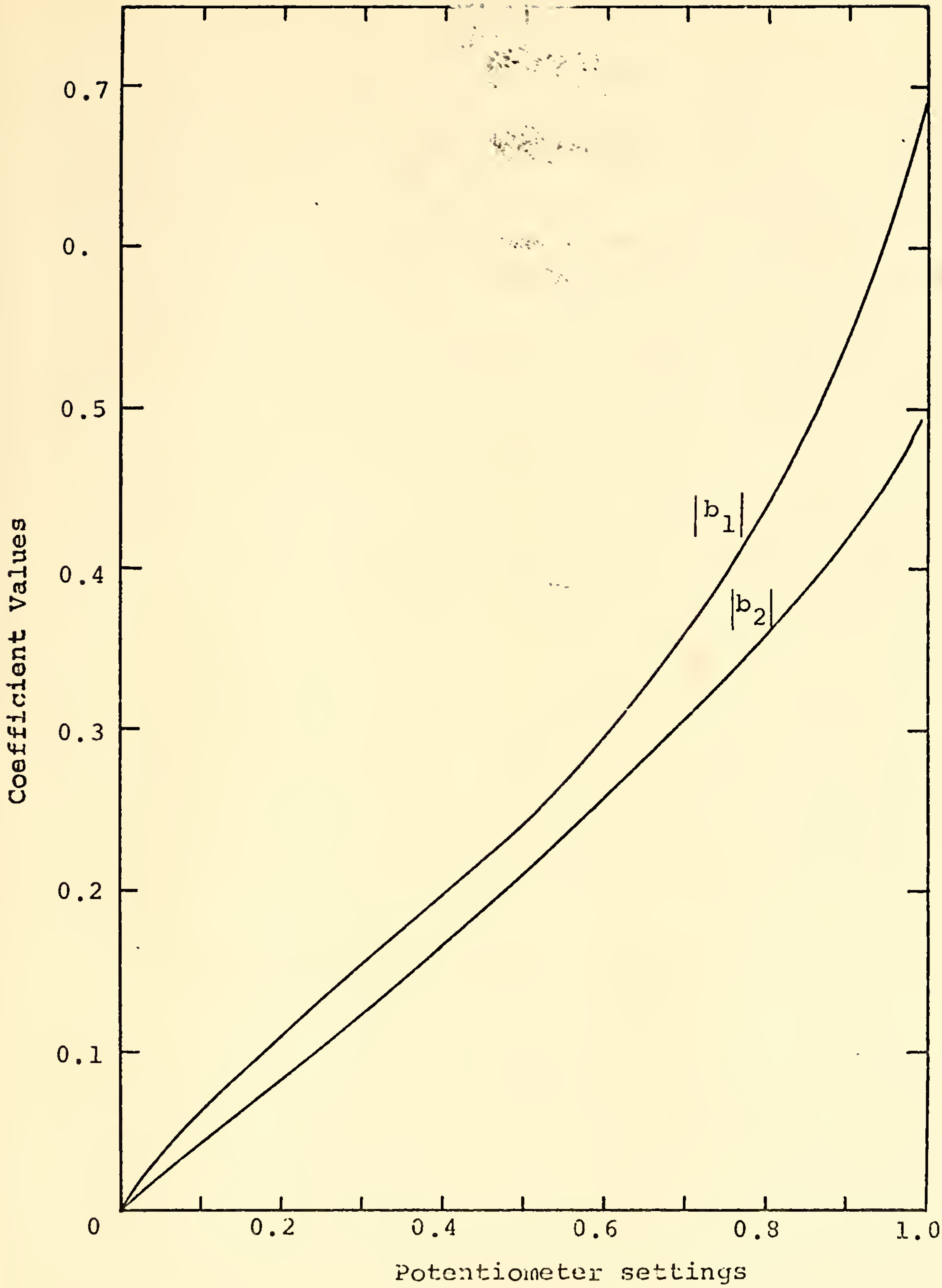


Figure 5.1 Calibration Chart for Coefficients Setting

VI. CONCLUSION

With the electronic industry achieving great progress in applying CCD technology in the fields of imaging, semiconductor memory, and transversal filtering for signal processing, it is noted that, in filtering, the CCD technology offers the low power consumption property of the analog filter and the high functional density and the stable low frequency performance of the digital approach. The results of this thesis project indicates that CCD recursive filtering will be applicable to many signal processing systems with the mentioned properties.

Comb-filters are very widely used in many signal processing systems, some of which are sonar, pulse-doppler radar, synthetic aperture radar, and vocoding system. A low-pass or a high-pass discrete analog recursive filter using n-bit CCD will provide the comb-filter needed for these purposes. The number of bits of CCD required needs to be at least twice as large as the number of frequency bins, and the clock frequency twice as high as the highest frequency of the expected signal. The CCD discrete analog recursive filter will become more attractive to these system applications as more research will be carried out on this subject.

The simplicity of the CCD discrete analog recursive filter offers the ease of frequency tuning and wide varieties of filters can be implemented with a second-order building block. These facts were demonstrated throughout the project. Therefore, it was expected that more applications of this type of filter will be found and investigated along with the advancement of the CCD technology. Some of the fabrication designs that would help the performance and the practicality of the CCD discrete analog recursive filter are on-chip arithmatic operations, non-destructive outputs, and on-chip sample-and-hold circuits, all of which would make a second-order CCD filter

building block more integrated and thus more compact. Reticon SAD-100's helped a great deal in predicting the performances of the second-order CCD discrete analog recursive filter since insufficient equipments did not allow experiments with more than one CCD at this time. The first-order CCD filter and both the first-order and second-order SAD-100 filters were very easy to work with, it was not expected that much difficulty or complication would be encountered with the second-order CCD discrete analog recursive filter experiments. However, some of the precautions which will make the experiments successful are to maintain proper quiescent voltage levels on gate c_1 and $gate_2$, to use low coefficient settings to avoid possible saturation since the input and output configurations of the CCD generally have a gain greater than unity, and to use only one input gate as the input port at first.

The objective of this thesis project was to investigate the feasibility of using CCD's in analog recursive filtering. It has been demonstrated that both the theoretical and the experiments were successful and encouraging.

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